Tieline®
The Codec Company

Merlin IP Codec
User Manual

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1 Warnings & Safety Information

1. Both appliance power cables must be removed from the device for Power Disconnection.
2. Remove the phone cable from the POTS interface before servicing.

THUNDERSTORM AND LIGHTNING WARNING:

DO NOT USE Tieline codecs during thunderstorms and lightning. You may suffer an injury using a phone, Tieline codec, or any device connected to a phone during a thunderstorm. This can lead to personal injury and in extreme cases may be fatal. Protective devices can be fitted to the line, however, due to the extremely high voltages and energy levels involved in lightning strikes, these devices may not offer protection to the users, or the Tieline codec and equipment connected to the codec.

Secondary strikes can occur. These secondary strikes are induced by lightning strikes and also produce dangerously high currents and energy levels. You only need to be near an object struck by lightning to lead to personal injury or damage to equipment. e.g. if you are located near a lighting tower at a sports facility, water features and drains on golf courses, you will be affected by these secondary strikes.

Damage to personnel and Tieline codecs may occur during thunderstorm, even if the codec is turned off but remains connected to the phone system or the power.

ANY DAMAGE TO A TIELINE PRODUCT CAUSED BY LIGHTNING or an ELECTRICAL STORM WILL VOID THE WARRANTY. Use of this product is subject to Tieline's SOFTWARE LICENSE and WARRANTY conditions, which should be viewed at www.tieline.com/support before using this product.

DIGITAL PHONE SYSTEM WARNING:

DO NOT CONNECT YOUR Tieline CODEC TO A DIGITAL PHONE SYSTEM. PERMANENT DAMAGE MAY OCCUR! If you are unfamiliar with any facility, check that the line you are using is NOT a digital line. If the Tieline codec becomes faulty due to the use of a digital phone system, the WARRANTY WILL BE VOID.

WARNING:

HIGH LEAKAGE CURRENT. EARTH CONNECTION ESSENTIAL BEFORE CONNECTING SUPPLY.

If the total leakage current exceeds 3.5 mA, or if the leakage current of the connected loads is unknown, connect the supplementary ground terminal to a reliable ground connection in your facility.

Supplementary ground connection

A supplementary ground terminal is provided on the codec to connect the unit to a ground connection. The ground terminal has an M4 stud with M4 retaining nuts and is compatible with all grounding wires. Remove only **NUT** 2 to connect your ground wire. The ground wire must have a suitable lug. When refitting **NUT** 2 ensure that both **NUT** 1 & **NUT** 2 are correctly tightened to establish and maintain a proper earth connection.

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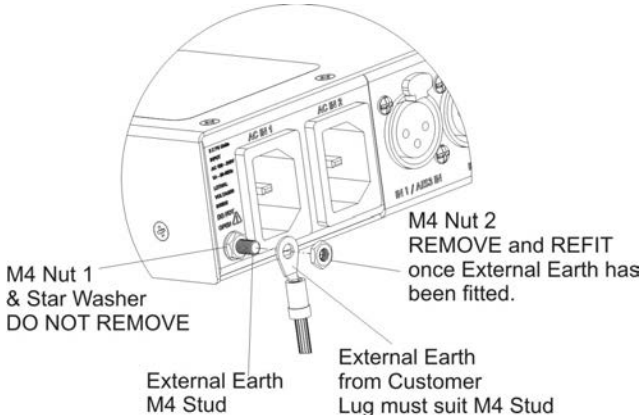

Disclaimer

Whilst every effort has been made to ensure the accuracy of this manual we are not responsible for any errors or omissions within it. The product specifications and descriptions within this manual will be subject to improvements and modifications over time without notice, as changes to software and hardware are implemented.

2 How to Use the Documentation

Overview of this User Manual

Use this manual to learn how to:

- Connect your codec to via a network interface and configure connections.
- Adjust audio and other settings within the codec.

Please read Getting Connected Quickly for an overview of how to configure your codec using 'programs' to store connection settings.

Manual Conventions

Warnings: Instructions that, if ignored, could result in death or serious personal injury caused by dangerous voltages or incorrect operation of the equipment. These must be observed for safe operation.

Cautions: Instructions warning against potential hazards, or to detail practices that must be observed for safe operation and to prevent damage to equipment or personnel.

Important Note: Information you should know to connect and operate your codec successfully.

Typographic Conventions

- Codec software elements are in Arial bold, e.g. **Contacts**
- Codec hardware elements are in bold Capitals, e.g. **KEYPAD**

Help Button

Press the [i] (information/help) button when navigating codec menus to display a dialog suggesting the actions which can be performed from within the current menu.

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3 Glossary of Terms

AES/EBU	Digital audio standard used to carry digital audio signals between devices.
AES3	Official term for the audio standard referred to often as AES/EBU.
DNS	The Domain Name System (DNS) is used to assign domain names to IP addresses over the World-Wide Web.
Domain	A group of computers or devices on a network which are administered with common rules and procedures. Devices sharing a common part of the IP address are said to be in the same domain.
DSCP	The Differentiated Services Code Point is a field in an IP packet header for prioritising data when traversing IP networks
Fail over	Method of switching to an alternative audio stream if the primary connection is lost.
GUI	Graphical User Interface
ISP	Internet Service Providers (ISPs) are companies that offer customers access to the internet
IP	Internet Protocol; used for sending data across packet-switched networks.
Latency	Delay associated with IP networks and caused by algorithmic, transport and buffering delays.
Multicast	Efficient one to many streaming of IP audio using multicast IP addressing.
Network Address Translation (NAT)	A system for forwarding data packets to different private IP network addresses that reside behind a single public IP address.
Port Address Translation (PAT)	Related to NAT; a feature of a network device that allows IP packets to be routed to specific ports of devices communicating between public and private IP networks.
PSU	Power Supply Unit
QoS (Quality of Service)	Priority given to different users or data flows across managed IP networks. This generally requires a Service Level Agreement (SLA) with a Telco or ISP.
RTP	A standardized packet format for sending audio and video data streams and ensures consistency in the delivery order of voice data packets.
SDP	SDP defines the type of audio coding used within an RTP media stream. It works with a number of other protocols to establishes a device's location, determines its availability, negotiates call features and participants and adjusts session management features.
SIP	SIP is a common protocol which works with a myriad of other protocols to establish connections with other devices to provide interoperability.
SLA	Service Level Agreements (SLAs) a contractual agreement between an ISP and a customer defining expected performance levels over a network
SNMP	Simple Network Management Protocol
STL	Studio to transmitter link for program audio feeds.
STS	Studio to studio audio link
TCP	TCP protocol ensures reliable in-order delivery of data packets between a sender and a receiver.
UDP	The most commonly used protocol for sending internet audio and video streams. UDP packets include information which allows them to travel independently of previous or future packets in a data stream.
Unicast	Broadcasting of a single stream of data between two points.

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4 Getting to know Merlin

The 1RU Merlin rack mount IP codec is designed for the latest digital IP broadcast networks and delivers high quality bidirectional stereo and full duplex communications for remote broadcast connections. Merlin will connect with all Tieline IP audio codecs and has multiple levels of power, audio and network redundancy.

Merlin

Applications

Built upon the success of the renowned Commander G3 rack mount codec, Merlin delivers high quality bidirectional stereo audio and full duplex communications for remote broadcast connections over LANs, WANs, the internet and satellite IP. It is ideal for studio and remote truck installations and features:

- High reliability over IP networks without Quality of Service.
- Powerful audio and data routing.
- Simple local or remote command and control.
- Recallable connection programs.

Codec Features

- DSP-based architecture designed for 24/7/365 operation.
- Dual Gigabit (10/100/1000) Ethernet ports with automatic switching for redundancy.
- Auto switching, dual redundant AC power supplies.
- Stereo peer-to-peer and point-to-point audio.
- Uncompressed PCM audio plus the low-delay, cascade resilient E-apt-X algorithm (capable of up to 24bit, 48kHz audio sampling)
- Other popular algorithms including LC-AAC, HE-AAC v1 and v2, MPEG-1 Layer II and III, Tieline Music and MusicPLUS, G.722 and G.711.
- SmartStream IP technology for automatically managing connections over IP networks.
- IPv4 & IPv6 compatible and ready.
- Asymmetric algorithmic encode/decode*.
- Integrated alarm management.
- G5 Toolbox GUI enables remote codec control over WANs.
- Compatible with Tieline Codec Management System**.
- Low latency in-band RS-232 aux data channel.
- Programmable software rules engine via a GUI for Control Port functions.
- Streamlined codec wizards and GUI for configuration and control.
- *Support for multiple languages: English, Spanish, German, Portuguese, French and Chinese.
- Connect to all Tieline IP codecs and Report-IT Live Enterprise Edition.

* Supported in later releases.
** Separate product

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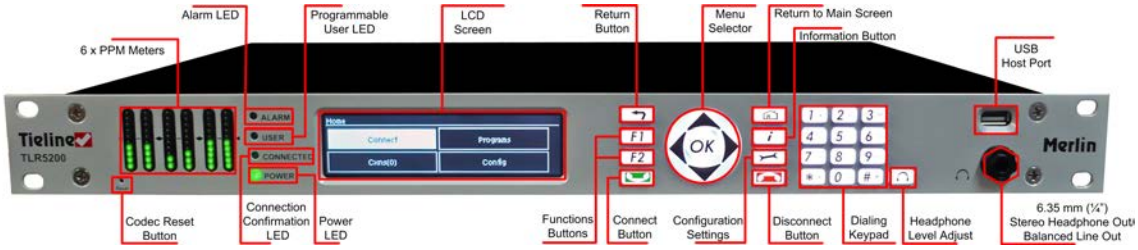
5 Merlin Front Panel Controls

The hardware front panel interface features menu navigation buttons, an LCD display with PPM metering and a dialing keypad.



Navigation Buttons

The codec has four arrow shaped navigation buttons for navigating codec menus and adjusting levels, and an **OK** button for selecting menu items.

Dialing Keypad

The keypad has alpha-numeric buttons, plus star and hatch (pound) buttons, which can be used to enter contact and program information into the codec.

Operation Button Descriptions

	Features	Operation Button Descriptions
	Return Button	Press to move back through menus & delete characters
F1	Function Button 1	Press to activate codec user functions
F2	Function Button 2	Press to activate codec user functions
	Connect Button	Press to create an IP connection
	Home Button	Press to return to home screen
	Information Button	Press to view a help menu onscreen
	Configuration Button	Press to configure codec settings
	Disconnect Button	Press to end a connection
	Headphone Button	Press to adjust headphone audio levels
	Reset Button	Press to reboot the codec

Adjusting LCD Screen Contrast Levels

1. Press and hold the button and then press the arrow up button to display the **Contrast** adjustment screen.
2. Use the left and right arrow buttons to adjust the LCD screen contrast until viewing is optimised.
3. Press OK when you have finished.

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Contrast can also be adjusted by pressing the **HOME** button, selecting **Config**, then **System**, and using the down button to navigate to **Contrast**.

Stereo RTS Headphone Output

The codec has a 6.35mm (1/4") RTS stereo **HEADPHONE** output for audio monitoring and this can also be switched to a balanced mono auxiliary line output. The front panel **HEADPHONE** output and rear panel **HP/AUX OUT** share the same hardware output. This means both are switched and configured together. I.e. both outputs are either a stereo headphone output (default setting), or a balanced mono auxiliary output.

USB 2.0 Host Port

USB 2.0 host port, which can be used for playback of backup audio files and firmware upgrades.

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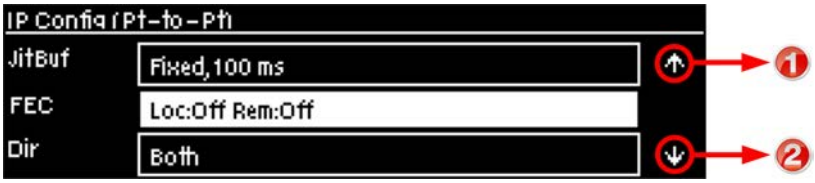
6 Navigating Menus

All main codec menus can be launched from the **Home** screen which includes:

	Features	Codec Home Screen Elements
1	Screen Name	The name of the current screen
2	Connect	Select to connect and adjust connection settings
3	Cxns	Displays the number of current connections and connection details
4	Programs	View and edit Program configurations
5	Config	Select to configure codec settings

Press the **RETURN** button to navigate backwards through menus, or press the **HOME** button to return to the **Home** screen from any menu.

If a complete menu cannot be viewed on a single codec screen, arrows on the right hand side of the screen indicate that the current menu has options below and/or above the visible items. Use the navigation arrows to scroll up and down.



	Features	Codec Home Screen Elements
1	Up Arrow	Arrow indicating menus can scroll upwards
2	Down Arrow	Arrow indicating menus can scroll downwards

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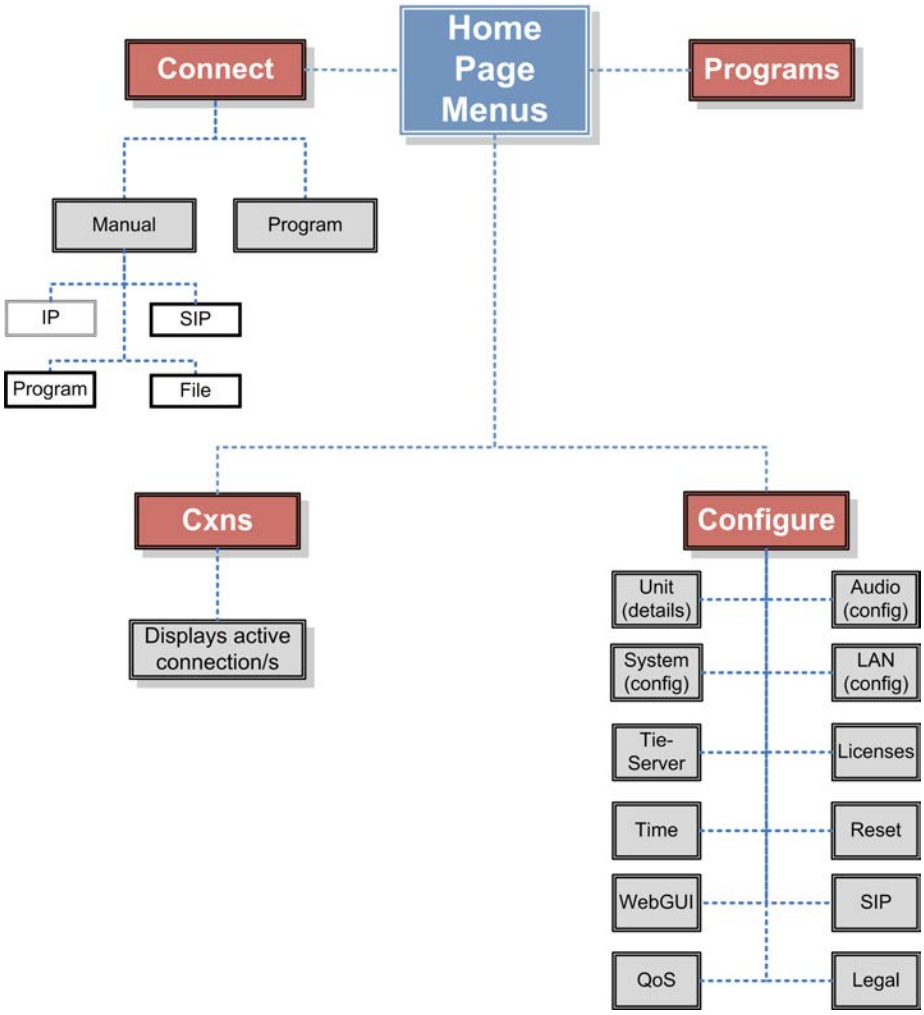
Codec Menu Overview

Following is an overview of the codec menus. The **Connect** and **Configure** selections on the main screen provide a range of configuration settings.



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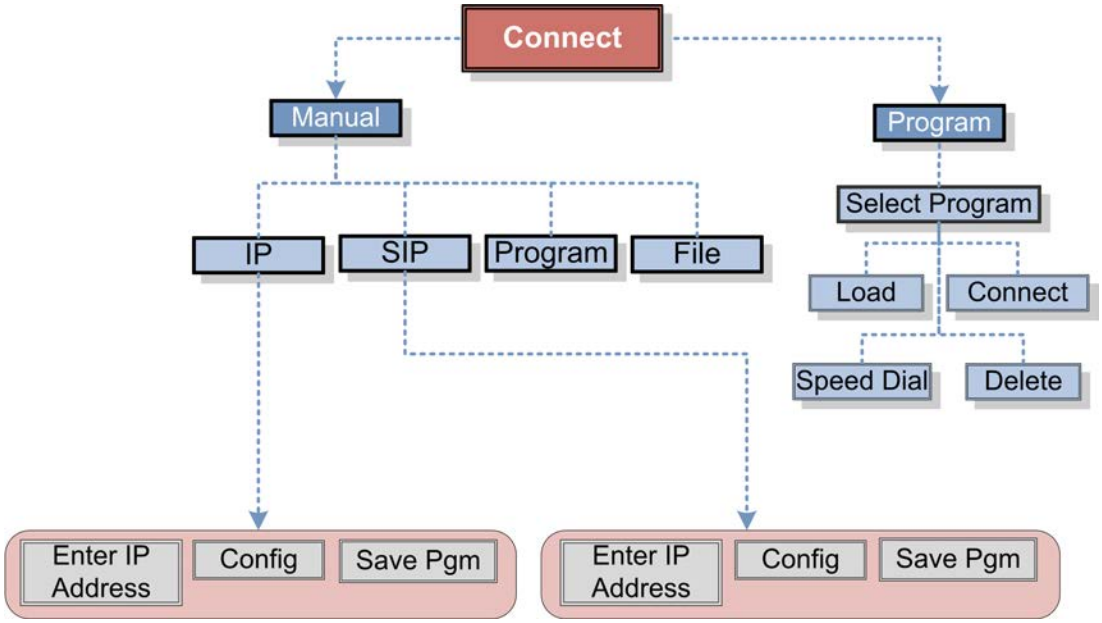
Connect Menu



Important Note: Menus for Multicast Server and Multicast Client modes of operation are only available for Genie Distribution IP audio codecs.

After selecting either **IP** or **SIP** in the **Connect** menu, select the **Config** menu to adjust configuration settings for the connection.

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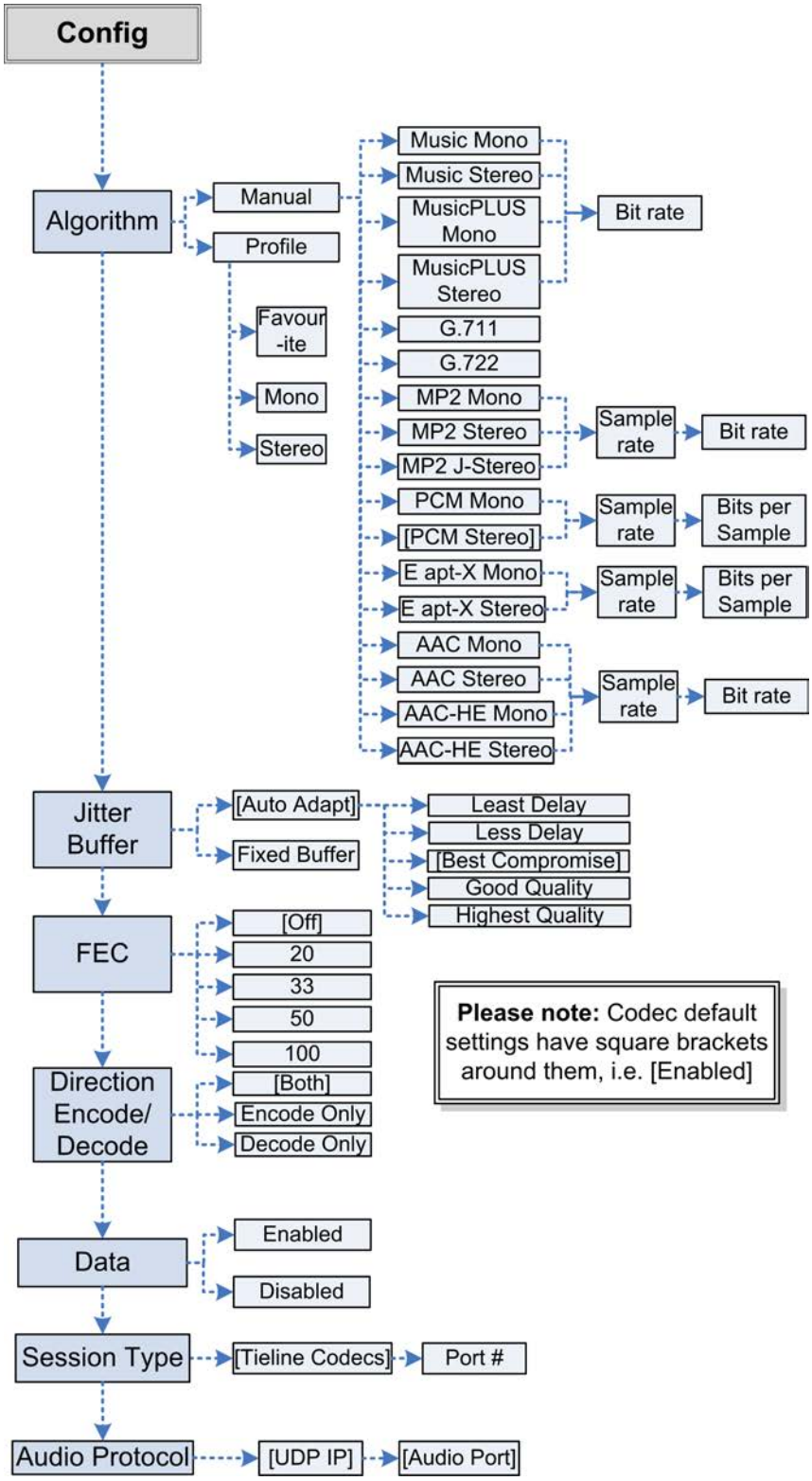

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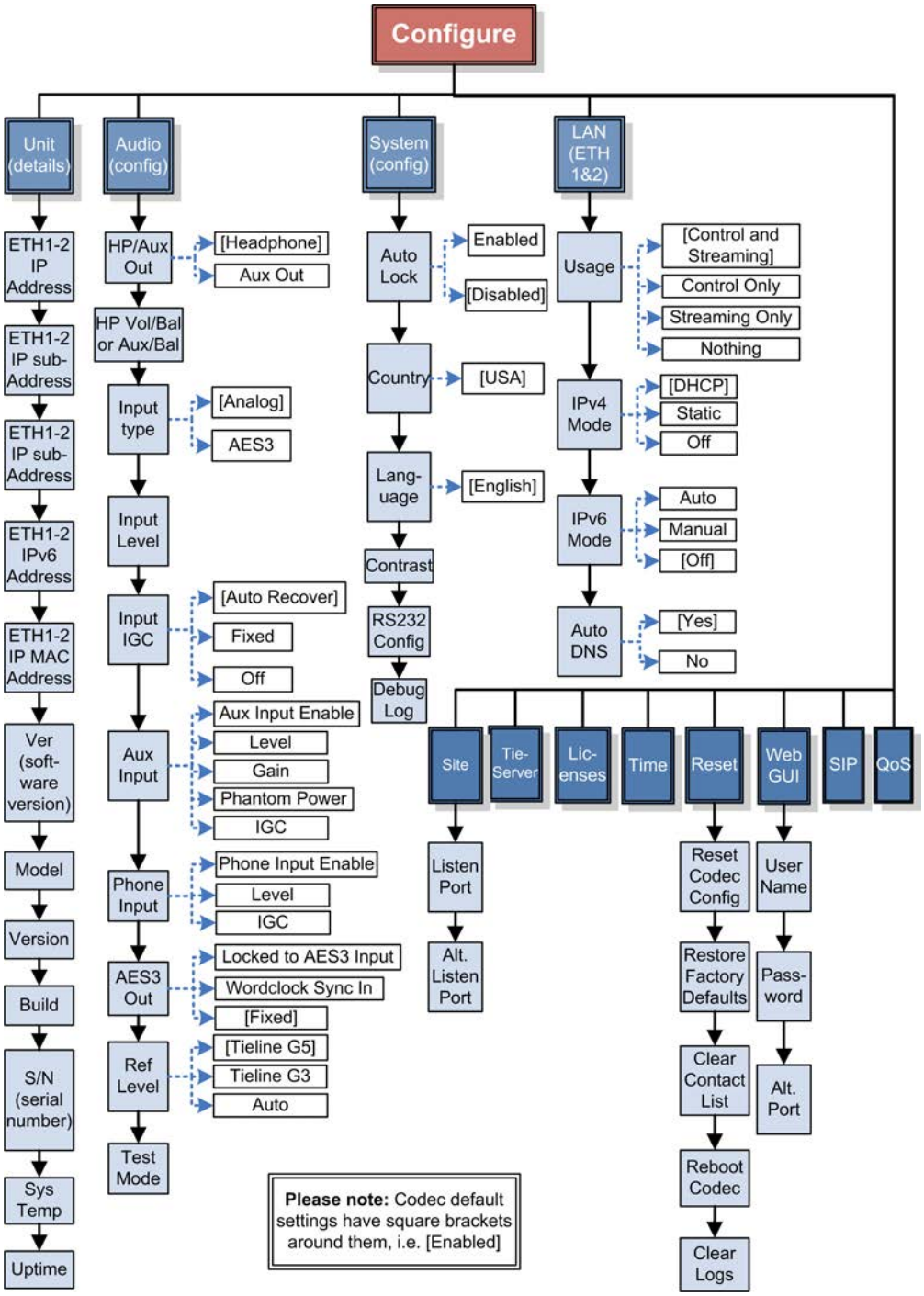
Configure Menu



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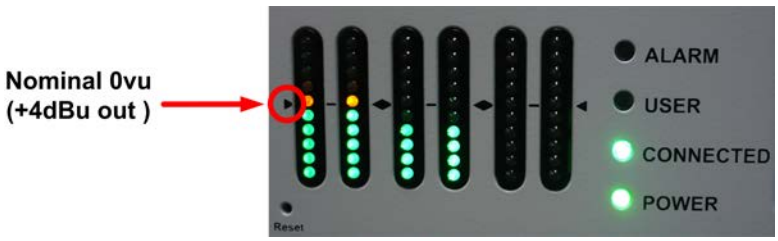
7 Merlin Input Levels and PPMs

The PPM meters use dBu to express nominal operating, headroom and noise floor levels.

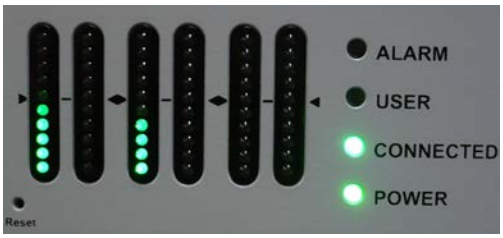
Audio Levels and Meters

Set audio levels so that audio peaks average at the nominal 0vu point indicated on the front panel PPM meters. This represents a program level of +4 dBu leaving the codec. Audio peaks can safely reach +22 dBu without clipping, providing 18dBu of headroom from the nominal 0vu point.



Mono and Stereo Metering

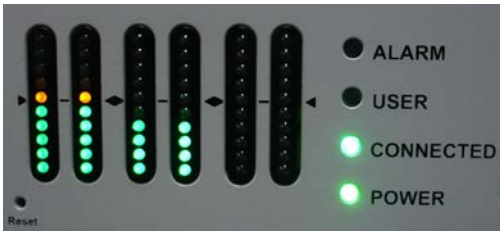
When connected with a mono program the codec will display audio on **PPM1** for input 1 and **PPM 3** for the return audio.



Mono connection displaying audio on PPMs 1 and 3

Important Note: If audio is present on inputs 1 & 2 the **PPMs** will display both channels before connecting, then audio will be summed and displayed on PPM1 after connecting.

When connecting with a stereo program, the codec displays audio on **PPM1** and **2** for inputs 1 and 2 and **PPM 3** and **4** for the return audio.



Stereo connection with PPMs 1-4 displaying input and return audio

Dual Mono Metering

Merlin is capable using a dual mono program configured to create two independent audio stream connections simultaneously. In this situation the codec will display connection 1 audio on **PPM1** for input 1 and use **PPM 2** for the return audio. Connection 2 audio will be displayed on **PPM3** for input 2 and **PPM 4** for the return audio.

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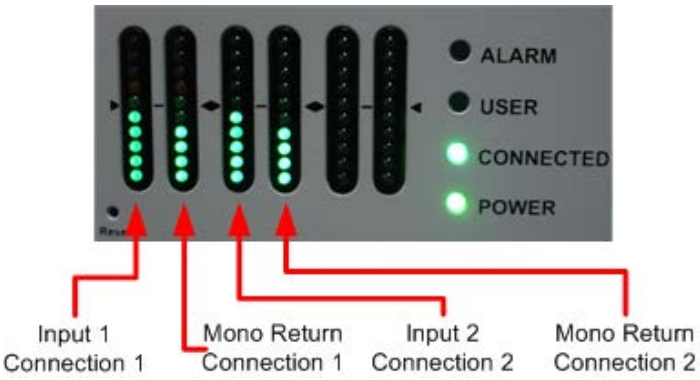
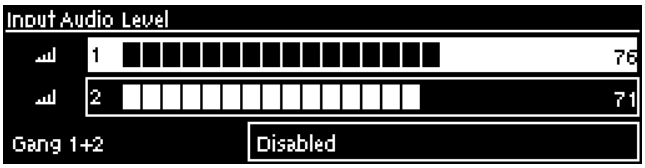


Adjusting Analog Input Levels

Important Note: Input levels can only be adjusted on analog inputs. Digital AES3 source audio is not adjustable. See Configuring AES3 audio for more information about the digital inputs and outputs. Input audio functions can be configured using the Toolbox web-GUI.

1. Press the **CONFIGURE** button.
2. Navigate to **Audio** and press OK.
3. Inputs are grouped in pairs under **Input Type** and should be set to **Analog**; press OK to toggle between **Analog** and **AES3** and press the **RETURN** button to exit the menu.
4. Use the down navigation button to highlight **Input Level** and press the OK button.
5. Navigate to the channels you want to adjust and press OK.
6. Use the left or right navigation buttons to select the appropriate gain setting, then press the OK button to save the settings.



Important Note: 15 volt phantom power can only be supplied on the Auxiliary input; this is disabled by default.

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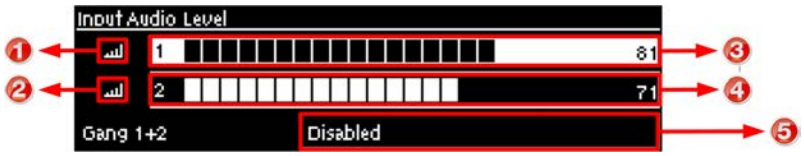
Quick Level Adjustment of Input 1 and 2

1. Press the F1 and right arrow buttons simultaneously to open the **Input Audio Level** adjustment screen.
2. Press 1 on the numeric keypad to toggle channel 1 on and off and press 2 to toggle channel 2 on and off.
3. Use the up and down arrow buttons to navigate to the channel you want to adjust. Note: A channel is highlighted when selected.
4. Use the left and right arrow buttons to adjust the input levels up or down.
5. Press the **RETURN** button to exit the screen.

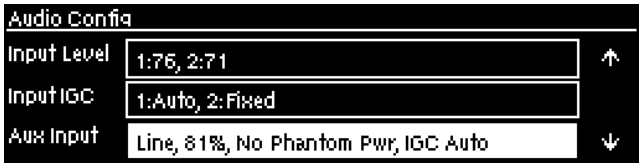


	Input Audio Features	Description
1	Channel On Symbol	Symbol indicates a channel is turned on
2	Channel Off Symbol	Symbol indicates a channel is turned off
3	Input 1 Level Control	Ch 1 level indication with percentage of gain indicated, i.e. **81**
4	Input 2 Level Control	Ch 1 level indication with percentage of gain indicated, i.e. **71**
5	Ch1/2 Gang Indication	Indicates whether ganging is enabled or disabled

Auxiliary Input Adjustment

The codec has 1 x 6.35mm (1/4") Mic/Line level Jack on the rear panel. By default the input is **Off** and can be configured by:

1. Selecting the **CONFIGURE** button.
2. Navigate to **Audio** and press the OK button**.**
3. Use the arrow-down button to select **Aux Input** and press the OK button to view menu options.



Input settings which can be adjusted include:

- Input on/off.
- Input level.
- Input Type: Unbalanced/Mic/Line
- Phantom power (15V available when enabled).
- IGC.

Important Note: When the auxiliary input (**AUX IN**) is **On** the default mixer configuration sends audio to inputs 1 and 2 for two channel program configurations and to input 1 for a mono program. Auxiliary input audio is sent to both inputs 1 and 2 for dual mono programs. If you are not using the Aux input ensure it is **Off** to avoid additional noise in program audio.

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Ganging Audio Channels

It is possible to gang channels together and adjust the audio level of the ganged channels simultaneously. When channels are ganged together:

- Both channels highlight together when selected.
- The gain setting for both channels is automatically set to match the gain level of the lowest of the two channels when ganging is first configured.
- If one channel is turned on when ganging is first configured then the other one will be turned on automatically.

1. Press the **CONFIGURE** button.
2. Navigate to **Audio** and press OK.
3. Use the down navigation button to highlight **Input Level** and press the OK button.
4. Navigate to the channels you want to gang and press the OK button.
5. Navigate to the **Gang** function and press the OK button to toggle between **Enabled** or **Disabled.**
6. Use the up and down arrow buttons to highlight and select the audio channels.
7. Use the left and right arrow buttons to adjust the levels for both inputs up or down simultaneously.
8. Press the **RETURN** or **HOME** buttons to exit the screen.

Important Note: If your codec supports using more than 2 inputs via a DB25 audio expansion connection, additional inputs are ganged in pairs, e.g. inputs 3 and 4, or 5 and 6 can be ganged together.

Quick Ganging of Audio Channels 1 and 2

Ganging is useful because it allows you to adjust the audio level of both inputs simultaneously.

1. Press the F1 button and the right arrow button to open the **Input Audio** level adjustment screen.
2. Use the up and down arrow buttons to navigate to and select **Gang 1 + 2 Enabled** or **Disabled.**
3. Press the OK button to select **Enabled**.
4. Use the up and down arrow buttons to highlight and select the audio channels.
5. Use the left and right arrow buttons to adjust the levels for both inputs up or down simultaneously.
6. Press the **RETURN** button to exit the screen.

Intelligent Gain Control (IGC)

When the broadcast action really starts to heat up, the codec's inbuilt DSP limiter automatically takes care of any instantaneous audio peaks that occur in demanding broadcast situations. **Input IGC** (Intelligent Gain Control) is enabled by default and is automatically activated at +20 dBu (G5 audio scale) and +14dBu (G3 audio scale) to prevent audio clipping.

There are three settings; **Auto**, **Fixed** and **Off**. If **Auto** is configured the codec will detect when incoming audio levels have reduced sufficiently and automatically return input levels to the gain setting prior to IGC being activated. The codec takes just 250 milliseconds to detect audio levels have returned to normal (after **IGC Level** has been initiated) and will return the levels to the previous setting within half a second. This response is linear.
To adjust this setting in the codec:

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1. Press the **CONFIGURE** button.
2. Navigate to **Audio** and press OK.
3. Navigate to **Input IGC** and press OK.
4. Select the channel you want to adjust and press OK.
5. Navigate to the preferred setting and press OK.

Programming Audio Metering when Connecting to Tieline G3 Codecs

New generation Genie, Merlin and Bridge-IT IP codecs have more audio headroom than Tieline G3 audio codecs, therefore metering needs to be adjusted when connecting to a Commander or i-Mix G3 codec with one of these codecs. The G3 metering scale is between -11dBu and +18dBu. Tieline codecs perform this metering adjustment automatically when they connect to each other or this can be programmed to occur by default.

1. Press the **CONFIGURE** button.
2. Navigate to **Audio** and press OK.
3. Navigate to **Ref Level** and press OK.
4. Select **Tieline G3** and press OK.

Audio levels should average around the nominal 0vu point and audio peaks should not exceed +16dbu as indicated by the PPM meter.

	Features	Description
1	-11dBu	PPM meter low point
2	+4dBu	Nominal 0vu reference level at +4dBu
3	+16dBu	+16 indication where audio will clip/distort

Important Note: If your codec (Genie Distribution and Bridge-IT) supports sending multi-unicast connections and the **Auto** (default) reference level is selected, the first codec you connect with will configure the reference level used for all subsequent multi-unicast connections.

Audio Metering when Connecting to Other Codecs

The **Other** setting is can be used for compatibility with other codecs. In particular it is used when connecting using G.711, G.722 and AAC algorithms.

1. Press the **CONFIGURE** button.
2. Navigate to **Audio** and press OK.
3. Navigate to **Ref Level** and press OK.
4. Select **Other** and press OK.

Audio levels should average around the nominal 0vu point and audio peaks should not exceed +12dbu as indicated by the PPM meter.

	Features	Description
1	-9dBu	PPM meter low point
2	+4dBu	Nominal 0vu reference level at +4dBu
3	+12dBu	+12 indication where audio will clip/distort

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8 Configuring AES3 Audio

If your input source is AES3 (AES/EBU) format use the **IN1/AES3** input on the rear panel of the codec. This is a balanced 110 ohm female XLR input and can operate effectively over distances of up to 100 meters. The input accepts both mono and stereo AES3 signals. Only a single XLR input or output is required to send and receive up to two channels of digital AES3 data.

1. Press the **CONFIGURE** button.
2. Navigate to **Audio** and press OK.
3. Select **Input Type** and press the OK button.
4. Navigate to the inputs you want to configure and press the OK button to toggle between **Analog** and **AES3**.

The 3 pin male XLR AES3 output on the rear panel is labeled **AES3 OUT**. It is capable of sending both mono and stereo AES3 signals.

Important Notes: Input levels are set at 100% automatically for AES3 connections. If you switch back to the analog input setting after selecting AES3, the previous analog settings will be recovered.

AES3 Sample Rate Conversion

The codec contains two sample rate converters.

Input Sample Rate Converter

The codec implements an Asynchronous Sample Rate Converter (ASRC) to convert the sample rate of the AES3 input to the sample rate set in the codec. The codec sample rate is determined by the selected algorithm. For example, if you select the Music algorithm, the sample rate will be set to 32kHz.

By default the codec will up-sample all channel 1 and 2 AES3 input sources to 96kHz sampling unless your audio source uses a 44.1kHz sample rate.

Output Sample Rate Converter

The sample rate of the AES3 output is currently configured using the clock source setting via the **CONFIGURE** button and then **Config > Audio > AES3 Out**. This configures the sample rate frequency of all AES3 output signals and there are three possible settings.

Locked to AES3 Input
If this setting is used, the codec will use the sync information received by the AES3 XLR input (this is the same as the **AES Rx Clock** setting in Tieline G3 codecs) to set the sample rate within the codec. This codec input also carries AES3 audio data.

Wordclock Sync In
This setting configures the codec for a word clock source via the **SYNC INPUT** on the codec rear panel (this is the same as the **External Word Clock** setting in Tieline G3 codecs). Often this will be a studio reference signal (D.A.R.S., or Digital Audio Reference Signal). In television

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broadcasting facilities, the audio reference signal should be locked to the video reference if there is one available. The sample rate being received is recognized by the codec and automatically adjusted within it. Sample rates from 32 kHz to 96 kHz are accepted, including the most popular rates of 32 kHz, 44.1 kHz and 48 kHz.

Fixed Sample Clock
Select from a range of fixed output sample rates.

Toolbox Web-GUI Output Sample Rate Configuration

The AES3 output sample rate can also be configured using the Toolbox web-GUI.

1. Open the web-GUI and click the **Configure symbol** at the top of the screen to open the **Configure panel**.

2. Click the **Audio tab** and use the drop-down menu to select your preferred **AES Output Clock** setting, then click **Save Settings**.

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9 Headphone/Aux Output

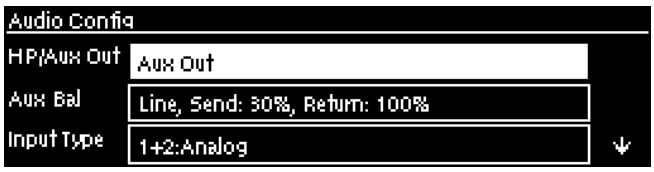
The codec has a 6.35mm (1/4") RTS stereo **HEADPHONE** output for monitoring inputs and return audio. If you are using analog inputs or digital inputs you will see audio metering on the PPMs and can monitor it with the headphones.

Important Note: The front panel **HEADPHONE** output and rear panel **HP/AUX** output share the same hardware output. This means both are switched and configured together. I.e. both outputs are either a stereo headphone output (default setting), or a balanced mono auxiliary output.

Configure for Headphone and Aux Output

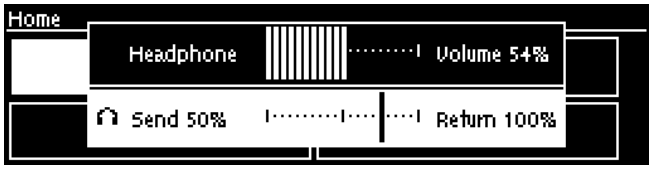
Both the front panel **HEADPHONE** and rear panel **HP/AUX** outputs are configured as stereo headphone outputs by default. To adjust this setting:

1. Press the **CONFIGURE** button.
2. Navigate to **Audio** and press OK.
3. Select HP/Aux Out and press OK to toggle between **Headphone** and **Aux Out**.



Adjust Headphone Output Settings

1. Press the **HEADPHONE** button to display the headphone monitoring adjustment screen.
2. Use the left ◀ or right ▶ navigation buttons to adjust the volume level up or down. The screen displays level adjustments in real-time.
3. Press the down ▼ navigation button to select the **Send/Return** audio balance and use the left ◀ or right ▶ navigation buttons to adjust the setting.



4. Press OK when you have finished.

The **Send/Return** audio balance dictates whether the front panel **HEADPHONE** output and the rear panel **HP/AUX** output monitors send (input/encoder) audio only, return audio only (decoder audio from a connected device), or a mix of both send and return audio.

Note: Headphone levels can also be adjusted by pressing the **SETTINGS** button, navigate to **Audio** and then **HP Vol/Bald** and press OK.

Adjust Auxiliary Output Settings

Settings for the auxiliary output audio are adjusted similarly to the **HEADPHONE** output, except that the output level is fixed at line level. Configure the front panel **HEADPHONE** output and rear panel **HP/AUX** output as an **Aux Out** and then:

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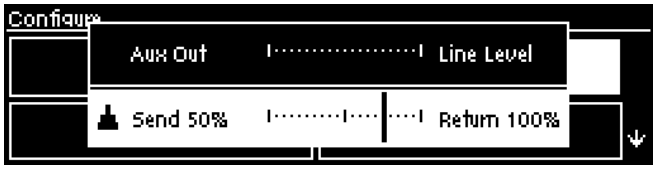
1. Press the **HEADPHONE** button to display the aux output adjustment screen.
2. Use the left ◀ or right ▶ navigation buttons to adjust the **Send/Return** audio balance.



3. Press OK when you have finished.

Note: Send/Return balance can also be adjusted by pressing the **SETTINGS** button, navigate to **Audio** and then **Aux Bal** and press OK.

10 Language Selection

English is the default language in the codec. To select a new language:

1. Press the **CONFIGURE** button.
2. Navigate to **System** and press OK.
3. Use the navigation buttons to select **Language** and press OK.
4. Select a language and press OK.

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11 About Program Dialing

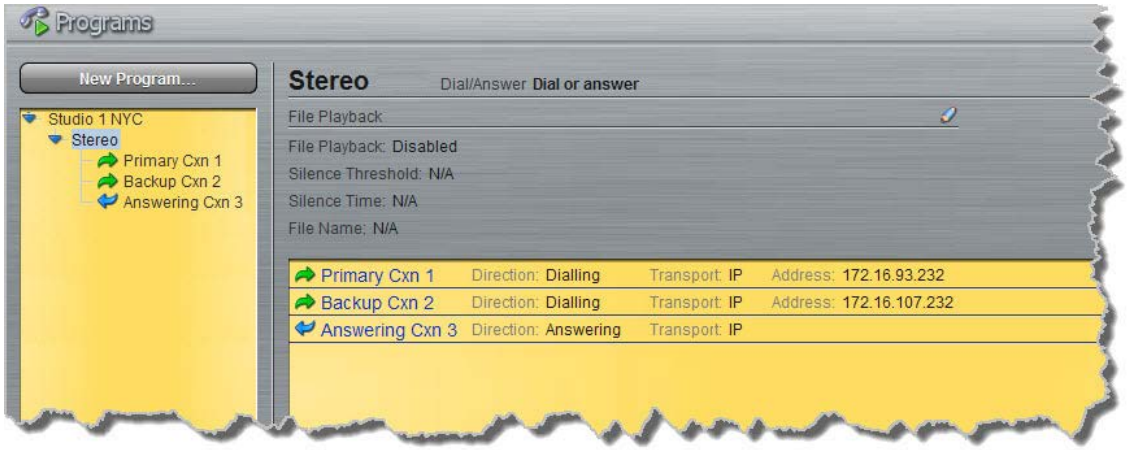
What Defines a Program?

A **Program** configures a Tieline codec to send or receive one or more **Audio Streams** based upon the particular application the codec is being used for at any given time.

Each audio stream within a program can be defined separately and contain a variety of settings relating to the number of connections (e.g. primary and backup) and the number of destinations to which each audio stream is distributed. Each audio stream is capable of being configured to be able to dial and answer connections, dial a connection only, or answer a connection only. Each one has its own:

- Name.
- Connection, Transport, and Destination settings.
- Backup configuration options.

The following image displays a simple point-to-point program in the **Programs panel** within the Toolbox web-GUI, which can be used to configure and edit all program parameters. The program displayed is configured to send a single stereo audio stream and will allow the codec to both answer and dial if required. It also has a backup dialing configuration in case the primary connection fails.



Creating Programs

Only the simplest point-to-point programs can be created using the codec front panel. The Toolbox web-GUI contains a feature-rich program creation wizard, which can tailor settings and create backup connections. Use the Toolbox web-GUI to retrieve or edit settings easily at the touch of a button. Once programs have been created they can also be used as templates for creating other programs using the web-GUI.

Important Note: Whether a connection is able to dial and answer connections will not affect its ability to encode or decode audio. Any type of connection is capable of bidirectional audio encoding and decoding.

Point-to-Point Programs (Unicasts)

New point-to-point programs can be created using the codec front panel keypad (see 10 Steps to Connect over IP). If you know the IP address of the codec you want to connect with then all you need to do is enter this into the codec, choose your preferred connection settings and then

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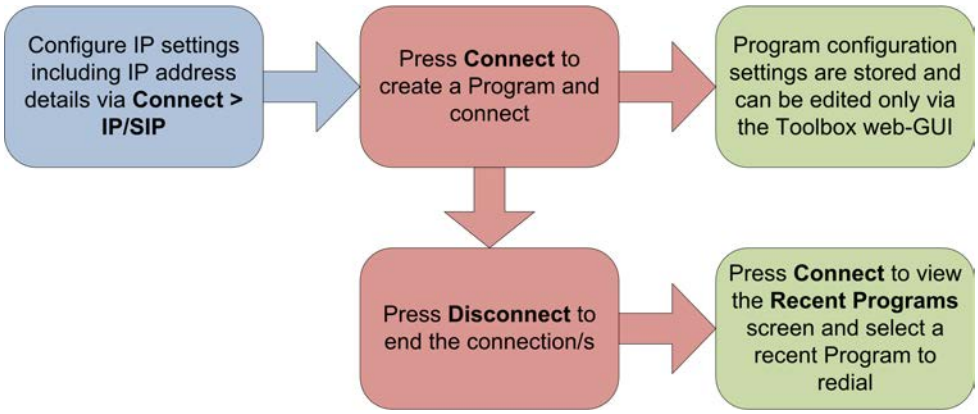
press **CONNECT** .

Front panel configured programs are automatically saved as **Recent Programs** - retaining all the audio stream dialing and configuration information programmed into the codec. These **Recent Programs** are displayed when you press the **CONNECT** button from within any menu except the **IP Mode** or **SIP Mode** screens, or the **Connect IP** or **Connect SIP** screens.

Ensure you configure all the correct connection settings first, as these are stored as part of the program's profile when you first connect. They cannot be adjusted afterward without using the editing features in the **Program panel** within the Toolbox web-GUI.

Important Note: When configuring a connection use the **Save** function in the **Connect IP** and **Connect SIP** screens to save programs permanently to the codec's **Programs** menu. Otherwise they are stored to the **Recent Programs** list and will be overwritten after several calls have been made.


Configure IP settings including IP address details via **Connect > IP/SIP**
Press **Connect** to create a Program and connect
Program configuration settings are stored and can be edited only via the Toolbox web-GUI
Press **Disconnect** to end the connection/s
Press **Connect** to view the **Recent Programs** screen and select a recent Program to redial

Point-to-point connection configured via the codec front panel

11.1 Multiple Stream Programs

Programs can be configured to connect a single audio stream to multiple destinations, or connect multiple audio streams to different destinations. The capability of your codec is dependent on the model you have purchased.

Dual Mono Programs

A dual mono program is made up of two audio streams. Each audio stream is a separate point-to-point connection with a different destination, which can also be configured with different transport, audio and backup settings. Each audio stream may also have its own backup connection.

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12 Getting Connected Quickly

Before attempting a new connection please connect and adjust the following:

1. Attach power to the codec.
2. Attach RJ45 Ethernet cables to at least one of the **ETH** ports on the codec's rear panel. IMPORTANT: Always use **ETH1** if you are connecting only one Ethernet cable to the codec. Programs created using the front panel of the codec are always configured to send audio over **ETH1**.
3. Attach headphones to the 6.35mm (1/4") headphone jack on the codec's front panel.
4. Check that the correct country is selected in the codec.

i. Press the **CONFIGURE** button.
ii. Navigate to **System** and press the OK button.
iii. Navigate to **Country** and press the OK button.
iv. Use the navigation buttons to select your country of operation and press the OK button.

5. Make sure you know the IP address of the destination codec.

12.1 10 Steps to Connect over IP

Important Note: The following procedure will create a custom program using the codec front panel keypad and navigation buttons. It pertains to connecting your codec point-to-point over IP for the very first time without using the Toolbox web-GUI. See Web-GUI Introduction for details on configuring connections remotely via a computer.

1. Press the F1 button and right navigation button to open the **Input Audio** level adjustment screen and adjust audio levels.
   - Press 1 on the numeric **KEYPAD** to toggle channel 1 on and off and press 2 to toggle channel 2 on and off.
   - Use the up and down navigation buttons to select **Gang 1 + 2** and press the OK button to toggle ganging on/off.
   - Use the up and down navigation buttons to select a single channel, or ganged channels. Note: A channel is highlighted when selected.
   - Use the left and right navigation buttons to adjust the input levels up or down.

2. Press the **HOME** button to return to the **Home** screen, select **Connect,** then select **IP** and press the OK button.

3. Use the **RETURN** button to delete numbers already entered, then use the numeric **KEYPAD** to enter the IP address of the codec you want to dial, using the * or # buttons to enter the periods in the IP address. Next, press the down navigation button to select **Config** and press OK.

Important Note: The codec remembers recent IP addresses just like a cell-phone. To view these addresses just press OK when you select the **Connect IP** screen. The most recent addresses and programs are listed first and you can use the navigation buttons to scroll up and down. Press OK to select the address you have highlighted.

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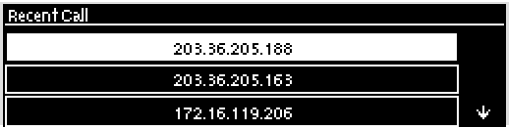
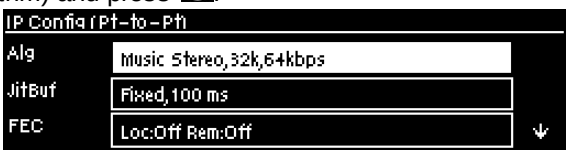
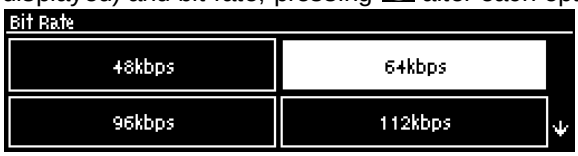


4. Navigate to **Alg** (algorithm) and press OK.

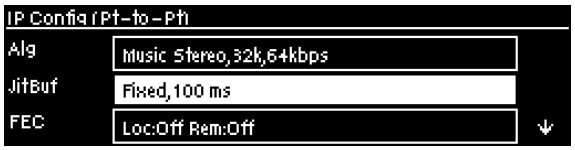


5. Use the navigation buttons to select an algorithm profile or manually enter algorithm settings, then press OK.

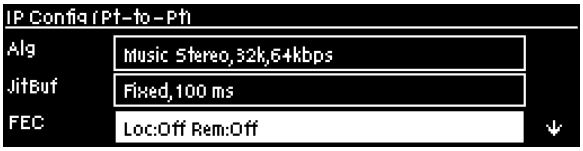
If you decide to manually program the algorithm, use the navigation buttons to select your preferred algorithm sample rate (if displayed) and bit rate, pressing OK after each option is selected.



6. Press the down navigation button to select Jitt (jitter buffer delay) and press OK to select a different automatic jitter buffer setting for your connection, or to enter a fixed buffer setting in milliseconds (maximum 999 ms). The default **Auto, Best Compromise** setting is a good starting point for most internet connections.



7. Press the down navigation button to select FEC (forward error correction) and press OK to view selection options. Use the navigation buttons to choose the FEC percentage you want to use and press OK.



8. When programming is complete press the **RETURN** button to navigate backwards to the **Connect IP** screen that the IP address was entered into.

Important Note: At this point you can navigate to **Save** on the **Connect IP** screen and press OK to use the numeric **KEYPAD** to name the program and press OK to save the program.

9. Press the **CONNECT** button to make a connection. The **Wait Connecting** screen appears during the connection process.

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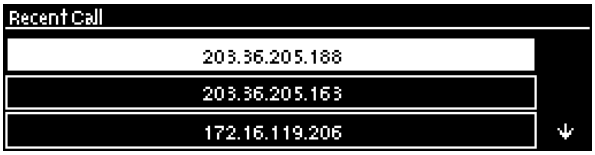

When connected, the green **CONNECTED LED** on the front of the unit will illuminate. Use the down navigation button to select **Cxns** and view connection **Status** and press OK to view connection statistics for IP packets being sent over the connection. To negotiate higher bit-rates press F2 then **3** on the numeric **KEYPAD**; for lower bit-rates press F2 then **9**.

12.2 Dialing Custom IP Programs

Custom programs are simple to dial from the codec front panel.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Programs** and press the OK button.
3. Use the up and down navigation buttons to select the program you want to connect with, then press the **CONNECT** button to make a connection.
4. The **Wait Connecting** screen appears during the connection process and then connection details are displayed.

12.3 Disconnecting a Connection

1. Press the red **DISCONNECT** button on the numeric **KEYPAD** at any time to hangup a connection.
2. Use the right navigation button to select **Yes** and press the **DISCONNECT** button or the OK button to confirm the disconnection.

12.4 Dialing SIP Peer-to-Peer

Important Note: When connecting to a Tieline G3 codec using SIP you need to manually select the G3 audio reference level. To do this select **CONFIGURE > Audio > Ref Level > Tieline G3**. In addition, select the following on the G3 codec prior to dialing.

- Select either a mono or stereo profile.
- Select **[Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP]**
- Select **[Menu] > [Configuration] > [IP1 Setup] > [Algorithm] > [G711/G722 or MP2]**

Dialing Peer-to-Peer IP Connections

SIP can be used to make direct peer-to-peer calls to different brands of IP codecs with public IP addresses, or between two codecs over a LAN which do not pass through firewalls. Peer-to-peer SIP calls are usually used to connect to other brands of codecs and perform call and session management tasks. Peer-to-peer SIP calls between two codecs are detected automatically and require no special pre-programming.

To make a peer-to-peer call between codecs we recommend both codecs use public IP addresses:

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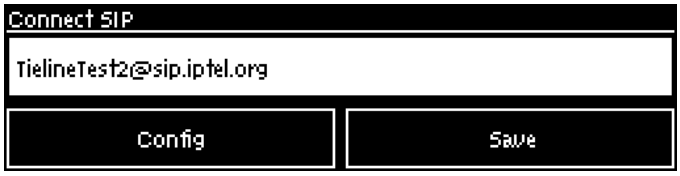
- Find out the IP address of the remote codec being dialed.
- Program each codec with a compatible algorithm and sample rate etc.
- Dial using **SIP** within the **Connect** menu.
- If the remote codec has a private IP address then it should be configured for port forwarding and should dial the public IP address at the studio (see Programming TCP/UDP Protocols for more details on port forwarding).

1. To dial peer-to-peer press the **HOME** button to return to the **Home** screen, select **Connect**, then select **SIP**.
2. Use the numeric **KEYPAD** to enter the IP address of the codec you want to dial, using the * or # buttons to enter the periods in the IP address and use the **RETURN** button to delete numbers already entered.
3. Then press the down navigation button to select **Config** and press OK to adjust the algorithm, jitter buffer and encode/decode direction if required.
4. Press the **RETURN** button to navigate backwards to the **Connect SIP** screen.
5. Press the **CONNECT** button to make a connection.

12.5 Dialing SIP Addresses

Dialing a SIP Address via the Codec Front Panel

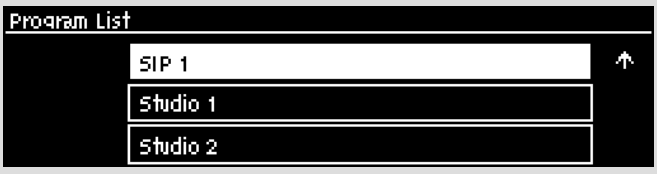
1. Press the **HOME** button to return to the **Home** screen, select **Connect,** then select **SIP** and press the OK button.
2. Use the **KEYPAD** to enter the numeric SIP address of the codec you want to dial, using the * or # buttons to enter the periods in the SIP address and use the **RETURN** button to delete any numbers already entered. Alternatively, if you have dialed the SIP address previously, press the **RETURN** button to view the **Recent Call** screen and select the SIP address you want.



3. Press the down navigation button to select **Config** and press OK, then adjust the algorithm, jitter buffer, encode/decode direction, port and auto reconnect settings if required.
4. Press the **RETURN** button to navigate backwards to the **Connect SIP** screen.
5. Press the **CONNECT** button to make a connection.

Important Notes: See Configuring SIP Settings for instructions on entering SIP account details into the codec.

After using the codec **KEYPAD** to dial the SIP connection for the first time, a new program is created. This will appear in the program list.



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It is also possible to configure SIP programs using the Toolbox web-GUI. See the section titled Configuring SIP Programs for more information.

12.6 Dial/Disconnect Multiple Audio Streams

Multiple Audio Streams within Programs

Some programs are created to allow simultaneous audio stream connections with different destination codecs, i.e. peer-to-peer dual mono programs. These programs can only be created using the Toolbox web-GUI.

There are two ways to simultaneously dial multiple audio stream connections within these types of programs:

1. Load the profile into the codec via the front panel and dial using the the **CONNECT** button.
2. Connect to the codec using the Toolbox web-GUI and use the Master panel to load the program and connect.

Dialing Multiple Audio Stream Programs with the Front Panel

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Programs** and press the OK button.
3. Use the up and down navigation buttons to select the program you want to connect with, then press the **CONNECT** button to make a connection.
4. The **Wait Connecting** screen appears briefly and then the **Home** screen is displayed.

It is also possible to redial the connection, see Redialing a Connection for more information.

Connection Details

Current audio stream connections are displayed on the **Home** screen via **Cxns**. In the following image two audio streams are currently connected.

Disconnect All Audio Stream Connections

1. Press the red **DISCONNECT** button on the numeric **KEYPAD** at any time to hangup all connections.
2. Use the right navigation button to select **Yes** and press the **DISCONNECT** button or the OK button to confirm the disconnection.

Disconnect a Single Audio Stream Connection

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Cxns** and press the OK button.

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3. Use the up and down navigation buttons to select the connection you want to disconnect.

3. Press the red **DISCONNECT** button on the numeric **KEYPAD**.
4. Use the right navigation button to select **Yes** and press the **DISCONNECT** button or the OK button to confirm the disconnection.

12.7 Monitoring IP Connections

The **Connected IP** screen displays information that assists in monitoring the performance of IP connections.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Cxns** and press the OK button.

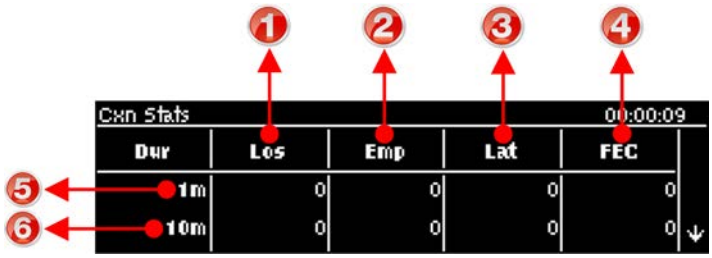
The IP address which has been dialed and the **LQ** (link quality) is displayed on the screen and you can use the down navigation button to view the algorithm being used, the connection bit-rate, total bytes used and the amount of jitter buffer delay over the IP network.

Link Quality (LQ) Readings

Local and Remote LQ numbers can also help you to determine if a problem is occurring at both ends of the connection or only one. For example, on an IP connection the "L" or "Local" reading represents the audio being downloaded from the network locally (i.e. audio data is being sent by the remote codec). Conversely, the "R" or "Remote" link quality reading represents the audio being downloaded by the remote codec (i.e. audio data is being sent by the local codec).

Viewing Connection Statistics

Navigate to **Status** in the **Connected IP** screen and press the OK button to display the **Cxn Stats** (connection statistics) screen. This displays the performance of the codec in sending IP audio packets across the network. Analysis is historic and assessed over 60 seconds and 10 minutes of connection time.



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	Feature	Description
1	Lost Packets	Packets sent and that failed to arrive
2	Empty (Jitter Buffer)	Indicates how often the jitter buffer 'reservoir' empties causing loss of audio
3	Late Packets	The number of packets that arrive late, i.e. after audio play out
4	FEC Packets	Indicates the number of forward error correction (FEC) packets that have been sent if it is enabled in the codec
5	1 minute	Statistics listed for the last minute of network activity
6	10 minutes	Statistics for the last 10 minutes of network activity

Important Note: If settings such as the jitter buffer, FEC or the connection bit rate are changed, it is best to assess a minute of connection performance in preference to 10 minutes of historical performance. This is because the 10 minute performance will include historical data, which includes connection settings that may no longer be relevant. It is only after a disconnection that 'packet arrival history' is cleared.

Following is a packet arrival analysis table with solutions for any noticeable packet loss statistics displayed on the screen.

Packet Analysis	Displays	Possible Causes	Possible Solutions
Loss	Packets sent and that failed to arrive.	• LAN/WAN congestion • Unreliable ISPs • Unreliable networks • Inferior IP hardware	• Renegotiate connection bit rate downwards • If link quality good add or increase FEC as required • Assess ISP's QoS if very bad performance
Empty	Indicates how often the jitter buffer 'reservoir' empties causing loss of audio.	• High number of packets being lost or arriving late • Signal dropouts using 3G cell networks • Renegotiation causes the jitter buffer reservoir to empty	• Once could be an anomaly – assess lost & late packets • If many lost packets and network is unreliable – renegotiate bit rate and /or FEC down • If many late packets, increase jitter buffer
Late	The number of packets that arrive late and after audio play out.	• Network congestion • Jitter Buffer depth is too low	• Auto-jitter buffer will adjust automatically • For manual jitter buffer settings increase jitter buffer depth 50-100 ms & reassess (if only a few packets arrive late over time, audio repairs will be automatic and may not require buffer changes).
FECd	Indicates the number of FEC repaired packets if FEC active.	• Packets have been lost or corrupted over the network	• Assess audio quality & the number of FEC repairs – if many packets are being 'lost' perhaps reduce FEC &/ or renegotiate bit rate down.

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12.8 Redialing a Connection

Press the **CONNECT** button from any codec menu to redial previous connections (except menus accessed via the **Connect** > **IP** screen).

Manually dialed connections are saved as programs - retaining all the dialing and configuration information programmed into the codec. A program is identified in the **Recent Program** redial screen using either a previously entered name, or by an IP address (manually dialed connections).

Redialing Manually from the Connect IP Screen

From the **Home** screen select **Connect > IP > Select an IP mode** and the codec assumes you want to dial a new manual connection. Press the **CONNECT** button when the **Connect IP** screen is displayed to retrieve previously dialed IP addresses. Codec settings used for a connection dialed from this screen include the current settings in the **Config** menu, which can be accessed via this screen.

12.9 Programming Auto Reconnect

Auto Reconnect is disabled by default. When enabled the dialing codec attempts to reconnect if audio is temporarily lost over an IP connection. This setting should only be programmed on the dialing codec to avoid dialing conflicts. To adjust the setting:

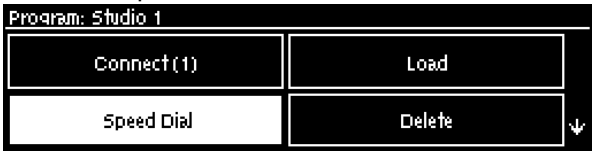
1. Press the **HOME** button to return to the **Home** screen, select **Connect,** then select **IP** and press the OK button.
2. Select the **IP mode** you are using to connect.
3. Select **Auto Recon** and press OK to toggle between **Disabled** and **Enabled**.
4. Select **Config** and press OK.
5. Navigate to **Auto Reconnect** and press OK to toggle between **Enabled** and **Disabled**.

Important Note: When **Auto Reconnect** is enabled, the dialing codec will continue to attempt a connection with the remote codec until **Disconnect** is pressed either on the dialing codec's keypad, or in the web-GUI.

12.10 Speed Dialing Connections

Assigning Speed Dial Numbers

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Programs** and press the OK button.
3. Navigate to the program you want to assign a speed number to and press the OK button.
4. Navigate to **Speed Dial** and press the OK button.



5. Navigate to the speed dial number you want to assign to the selected program and press the OK button.
6. A confirmation message will display the number assigned.

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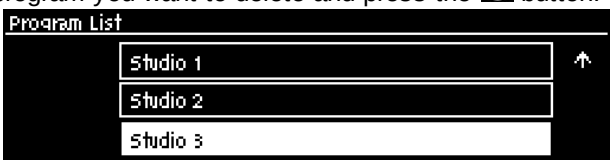
Speed Dialing

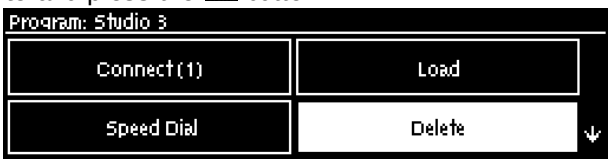
1. Press the **HOME** button to return to the **Home** screen.
2. Use the numeric **KEYPAD** to enter the speed dial number.
3. When the **Speed Dial** screen appears, press the OK button or the **CONNECT** button to connect.

12.11 Deleting Programs

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Programs** and press the OK button.
3. Navigate to the program you want to delete and press the OK button.



4. Navigate to **Delete** and press the OK button.



5. Confirm the deletion and press the OK button.

12.12 Choosing Dialing Profiles

A number of pre-programmed mono and stereo dialing profiles are available for programming the codec quickly without individually selecting algorithms and bit-rates etc. These profiles have been programmed with the most popular settings that provide high quality connections using each available algorithm.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Connect** and press the OK button.
3. Select **IP** and press the OK button.
4. Select your preferred **IP Mode** and press the OK button.
5. Use the down navigation button to select **Config** and press the OK button.
6. Select **Alg** and press the OK button.
7. Use the right navigation button to select **Profile**.
8. Choose the profile you want from the **Favorite**, **All**, **Mono or Stereo** menus.

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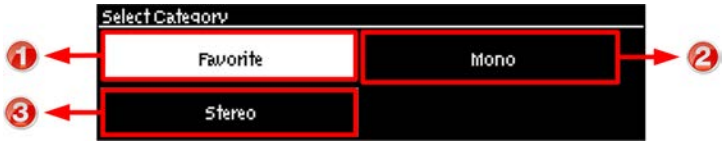
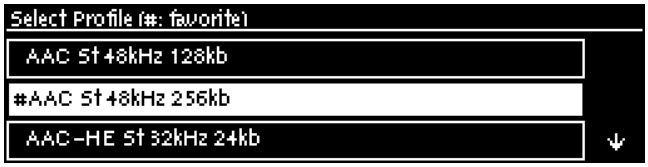


	Features	Codec Home Screen Elements
1	Favorite	Displays a list of favorite profiles that have been selected manually within the codec by users
2	Mono	Displays preprogrammed mono profiles within the codec
3	Stereo	Displays preprogrammed stereo profiles within the codec

Adding a Profile into the Favorite Menu

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Connect** and press the OK button.
3. Select **IP** and press the OK button.
4. Select your preferred **IP Mode** and press the OK button.
5. Use the down navigation button to select **Config** and press the OK button.
6. Press the OK button to select **Alg**.
7. Use the right navigation button to select **Profile**.
8. Select the profile you want from the **All**, **Mono or Stereo** menus.
9. Press the hatch button # to add the profile into the **Favorite** menu.

Profiles that have been added into the **Favorite** menu are identified by the hatch symbol next to their name after they have been selected.



Deleting a Profile from the Favorite Menu

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Connect** and press the OK button.
3. Select **IP** and press the OK button.
4. Use the down navigation button to select **Config** and press the OK button.
5. Press the OK button to select **Alg**.
6. Use the right navigation button to select **Profile**.
7. Select the profile you want to delete from the **Favorite** menus.
8. Press the hatch button # to delete the selected profile from the favorite menu.

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12.13 Merlin Profiles

The following algorithm profiles are pre-programmed into Merlin codecs.

	Profiles			
	Algorithm	**Mono/Stereo**	**Sample Rate (kHz)**	**Bit-rate (Kbps)**
1	AAC	Mono	48	64
2	AAC	Stereo	48	128
3	AAC	Stereo	48	256
4	HE-AAC	Mono	32	16
5	HE-AAC	Stereo	32	24
6	HE-AAC	Stereo	32	48
7	Enhanced apt-X	Mono	32 (16 bit)	128
8	Enhanced apt-X	Mono	48 (24 bit)	288
9	Enhanced apt-X	Stereo	32 (16 bit)	256
10	Enhanced apt-X	Stereo	48 (24 bit)	576
11	G.711	Mono	8	64
12	G.722	Mono	16	64
13	MPEG 1 Layer 2	J-Stereo	32	128
14	MPEG 1 Layer 2	J-Stereo	48	192
15	MPEG 1 Layer 2	Mono	24	64
16	MPEG 1 Layer 2	Mono	48	256
17	MPEG 1 Layer 2	Stereo	32	128
18	MPEG 1 Layer 2	Stereo	48	256
19	Music	Mono	32	28.8
20	Music	Mono	32	48
21	Music	Stereo	32	64
22	Music	Stereo	32	96
23	MusicPLUS	Mono	48	48
24	MusicPLUS	Mono	48	96
25	MusicPLUS	Stereo	48	96
26	MusicPLUS	Stereo	48	128
27	MusicPLUS	Stereo	48	192
28	PCM Mono	Mono	48 (16bit)	768
29	PCM Stereo	Stereo	48 (16bit)	1,540

12.14 Merlin Backup Connections

Tieline codecs feature comprehensive backup audio options which include automatic fail over to an alternative connection. For details on configuring backup connections using fail over see Configuring Merlin Point-to-Point Programs.

IP Backup Configuration

The codec features dual Ethernet IP ports allowing two completely independent IP connections. Currently the codec supports 'cold' backup whereby the second connection is dialed after the codec determines that the primary connection has failed.

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12.15 Locking the Front Panel

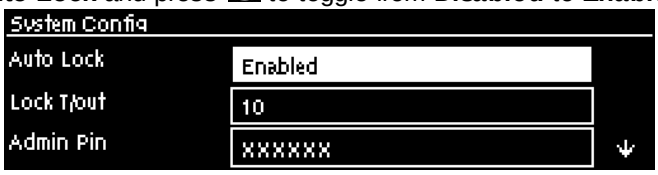
The codec features a front panel lock feature for tamper-proof operation. This feature is disabled by default.

There are two levels of panel lock and each requires a user to enter a PIN to access different features:

1. **Admin PIN**: Required to change codec Connection settings or configuration settings accessed via the **CONFIGURE** button. (Default PIN is: 456789)
2. **User PIN**: Required to use the codec front panel buttons and dial/hangup a connection (Default PIN is: 123456)

Enabling the Front Panel Lock Feature

1. Press the **CONFIGURE** button.
2. Navigate to **System** and press OK.
3. Navigate to **Auto Lock** and press OK to toggle from **Disabled** to **Enabled**.



4. Navigate to the panel **Lock Timeout** field and press OK to enter the desired time-out period in seconds.
5. If you want to change the default **Admin PIN** or **User PIN**, navigate to each in turn and press OK to enter a new PIN.

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13 Connecting to the Web-GUI

Codecs can be programmed using the Toolbox web-GUI and this can be launched using either an IP/LAN connection or a USB connection with the codec. Instructions for using the web-GUI are contained in the application itself from the **Help** panel. The Tieline web-GUI application runs on:

- Internet Explorer 6 or greater on Windows® XP, Windows Vista ® and Windows 7 ®.
- Firefox® 3 or greater on Windows® XP, Windows Vista ® and Windows 7 ®, Solaris™ and Linux®.

Web-GUI Prerequisites

1. To use the ToolBox web-GUI you will need to download the latest version of Java™ by visiting http://www.java.com. The Web-GUI will prompt you to do this if Java is not installed and you attempt to launch the ToolBox web-GUI.
2. After updating to the latest version of Java you need to refresh your browser.

13.1 Opening the Web-GUI & Login

1. Attach an Ethernet cable to the **Ethernet** port on codec.
2. Press the **CONFIGURE** button and select **Unit** to display the IP address programmed into your codec.
3. Ensure your PC is connected to the same LAN.
4. Open your web browser and type the IP address of your codec into the address bar of your browser, e.g. **http://192.168.0.xxx** (the last digits are the private address details unique to your codec over a private LAN).
5. Refresh the browser and the web-GUI application should launch automatically.
6. When you open the Toolbox web-GUI an authentication dialog prompts you to enter a password to login. The first time you log in you can enter the default setting "password" and click the **OK** button. Tieline highly recommends you click the hyperlink in the login dialog or visit Changing the Default Password to change the password and have greater security during live broadcasts.

Using the Web-GUI over the Internet

If your codec is connected over the internet via a public static IP address it is possible to connect and configure it from any PC which is also connected to the internet.

IPv6 Configuration

It is only possible to configure IPv6 connections using the Toolbox web-GUI.

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LAN Troubleshooting

PC LAN Settings

Check the LAN settings on your PC if it is connected to a LAN and is having trouble opening the Toolbox web-GUI in a web-browser.

1. Open Internet Explorer.
2. Click **Tools** and then click the **Connections** tab.
3. Click the **LAN settings** button.
4. If the PC is using a proxy server over the LAN you may need to select the **Bypass proxy server for local addresses** option box.
5. If you still can't connect, click the **Advanced** button in the **LAN Settings** dialog and ask your IT administrator to assist you with entering the IP address of the codec into the **Exceptions** pane of the **Proxy Settings** dialog.

Port Selection

By default port 80 is used by your PC to communicate with the codec and launch the web-GUI. If port 80 cannot be used across your network for some reason, type the IP address of your codec into your browser with a full colon and the port number 8080.

E.g. **192.168.0.176:8080**

It is also possible to specify a different port for connecting the Toolbox web-GUI to your codec.

1. Press the **HOME** button on the codec to return to the **Home** screen.
2. Use the navigation buttons to select **Config** and press the OK button.
3. Use the navigation button to navigate down to **WebGUI** and press the OK button.
4. Select **Alt. Port** and press OK.
5. Use the **KEYPAD** to enter a new port number and press the OK button to save the new setting (Note: there is no character limit for passwords).
6. Type the IP address of your codec into your browser with a full colon and then the new port number.

Important Note: Any new port specified must be within the range 2000 to 65535 inclusive.

Launching the Toolbox web-GUI

If you have trouble launching the web-GUI in a browser, type http://<insert codec IP address>.htm directly in your browser.

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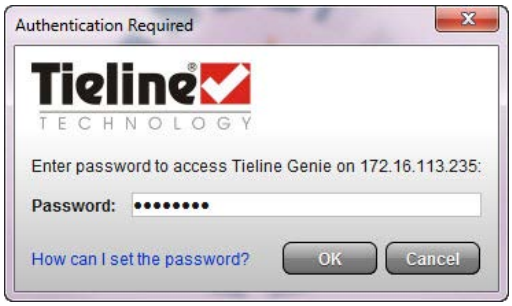
13.2 Changing the Default Password

The default password for the Toolbox web-GUI is **password**. This has to be entered to use the web-GUI and Tieline highly recommends changing the default password as soon as possible to protect your codec from being tampered with during live broadcasts.



Default Password Entered

Caution: Codecs connected to the internet can be accessed by anyone with knowledge of the codec's public IP address. Setting a strong password protects your equipment from being tampered with and jeopardizing live broadcasts.

Creating a New Password

The authentication login password can be changed at any time using the codec keypad and LCD screen. Note that passwords are case sensitive:

1. Press the **CONFIGURE** button.
2. Use the navigation button to select **WebGUI** and press the OK button.
3. Select **Password** and press OK.
4. Use the **KEYPAD** to enter a new password and press the OK button to save the new setting (Note: there is no character limit for passwords).

If you forget the password for the Toolbox web-GUI then you can always press the **CONFIGURE** button on the codec and navigate to **WebGUI** to view the current password and change it if required.

Important Note: The **Username** in the menu is permanently set to Tieline and cannot be changed; only the **Password** can be changed.

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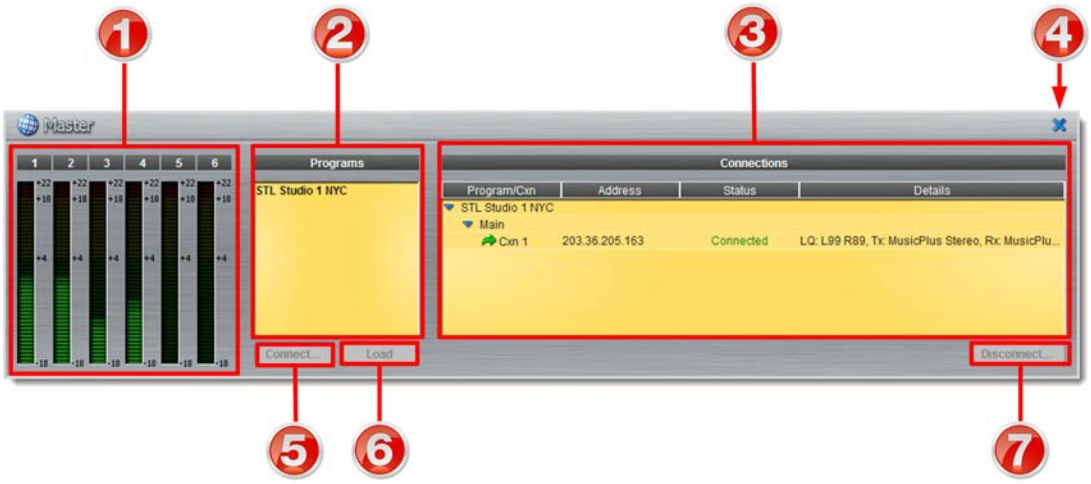
14 Using the Web-GUI

The following sections provide an overview of the different programming panels available within the codec's Toolbox web-GUI. Navigate with the mouse pointer to a symbol at the top of the web-GUI screen and click to open the panel selected. When a panel is opened in the web-GUI, the text below the symbol is highlighted (see **Master** in the following image).

Web-GUI Symbols for Opening Panels

Master Panel to Load and Connect Programs, Audio Streams and Connections



	Feature	Description
1	Input/Output PPMs	6 PPM meters to display audio levels for inputs and outputs
2	Programs list	Lists all configured programs which have been added into the codec. Click to select a program before loading or connecting
3	**Connections**	Provides a summary of connection details and audio streams
4	**Close** button	Click to close the **Master panel**
5	**Connect** button	Click **Connect** to connect all audio streams configured within the currently selected program in the **Programs** list; this button also loads the program currently selected in the **Programs** list
6	**Load** button	Click to **Load** the codec with the program currently selected in the **Programs** list
7	**Disconnect** button	Click to disconnect the currently selected audio stream or a specific connection. Note: this button becomes a **Connect** or **Unload** button when all audio streams are disconnected.

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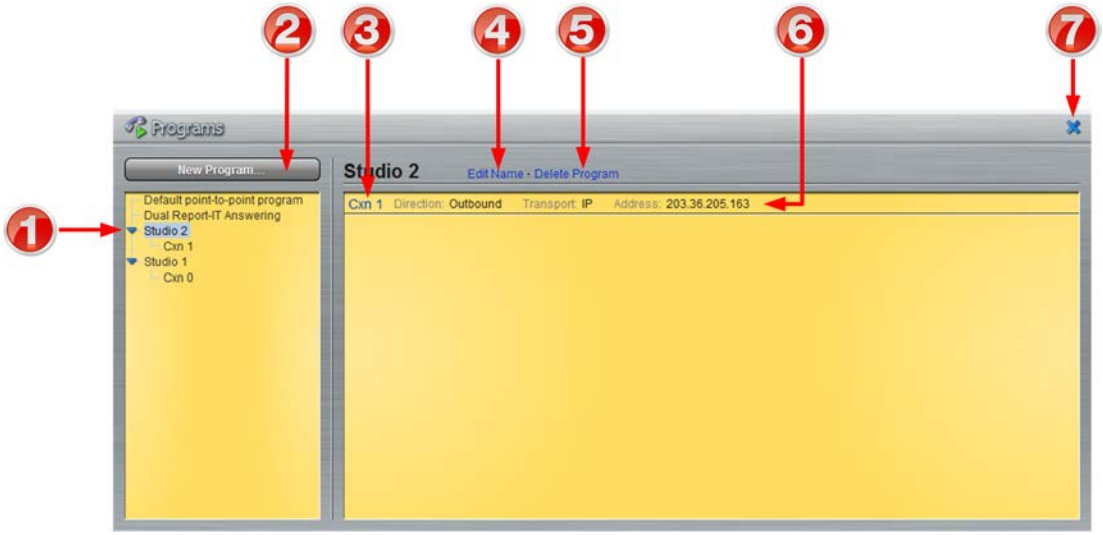
Programs Panel for Connection Configuration



	Feature	Description
1	Programs List	Displays the programs listed in the codec: Click the drop-down arrow to display connections within programs and click a connection to display its configuration settings. Displays program configuration details when creating a new program.
2	**New Program** button	Click to add a new program.
3	Connection Name	Click to reveal and edit all connection settings.
4	**Edit Name**	Click to edit the name of the currently selected program.
5	**Delete Program**	Click to delete the currently selected program.
6	Connection Overview	Brief connection overview
7	**Close** button	Click to close the **Connect panel**.

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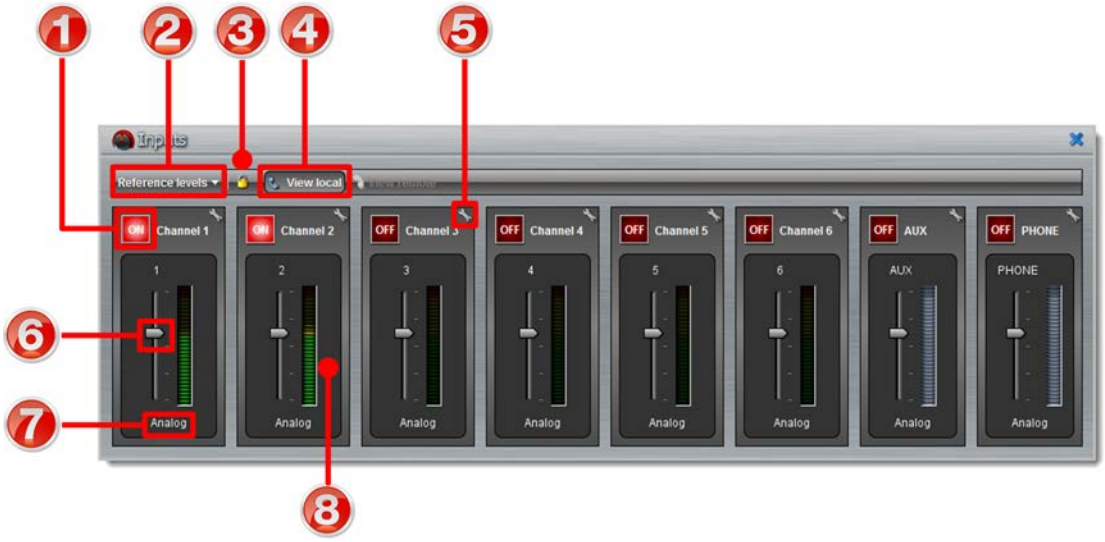
Inputs Panel for Input Adjustments



Important Note: Tieline codecs have different input configurations, therefore the image shown may not reflect the number of inputs displayed in your codec web-GUI.

	Feature	Description
1	Channel **ON/OFF** Buttons	Click to turn each channel **ON** or **OFF**
2	**Reference levels** menu	Click the drop-down arrow to select the codec input reference level (default setting **Auto**)
3	Lock Button	Click to lock all **Input** panel settings (greys out when locked)
4	**View local**	Click to view local codec inputs (default)
5	**Configure** button	Click to adjust input **Name**, **Type**, **IGC** and **Ganging**
6	Input Sliders/Faders	Input gain control sliders/faders
7	Analog/AES3 Indication	Indicates whether the codec input is configured for analog or digital audio sources
8	Input PPM meter	Input PPM meter

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Rules Panel for Creating Relay Activation Rules

Rules
What do you want this rule to do?
1
Connect and disconnect a program when an input is toggled
E.g. Connect "My Program" when Input 1 is switched ON, and disconnect it when Input 1 is switched OFF.
2
Connect a program when an input is switched ON and disconnect it when another input is switched ON
E.g. Connect "My Program" when Input 1 is switched ON, and disconnect it when Input 2 is switched ON.
3
Synchronise a local input to a remote relay
E.g. Set remote relay 1 to the value of input 1.
4
Toggle a relay based on a connection's status
E.g. Switch relay 1 ON when "My Program" is connected, and switch it OFF when "My Program" is disconnected.
Cancel

	Rule	Description
1	Connect/Disconnect a program by toggling a relay input	Click to program Connection and Disconnection by toggling an input
2	Connect when an input is switched **ON**; Disconnect when another input is switched **ON**	Click to program Connection and Disconnection after different relay inputs are switched **ON**
3	Synchronise a local relay input with a remote relay output	Click to program a local relay input to synchronise with the state of a remote relay output
4	Toggle a relay based on connection status	Click to program a relay to toggle based on connection status

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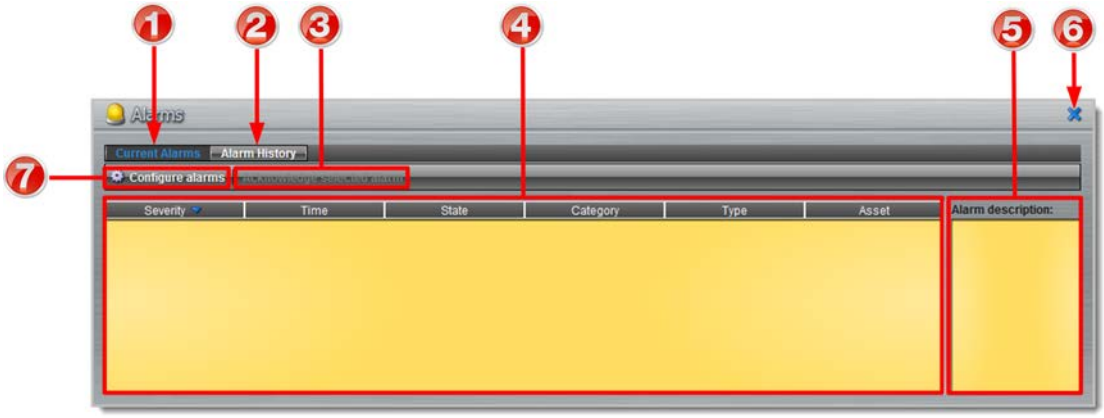
Alarms Panel



	Feature	Description
1	**Current Alarms**	Click to view current device alarms
2	**Alarm History**	Click to view the history of device alarms
3	**Acknowledge Selected Alarm**	Click to acknowledge an alarm after activation
4	**Alarm details** pane	Displays alarm details
5	**Alarm description** pane	Troubleshooting information to assist users when alarms occur
6	**Close** button	Click to close the **Alarms** panel.
7	**Configure alarms**	Click to create or edit alarms.

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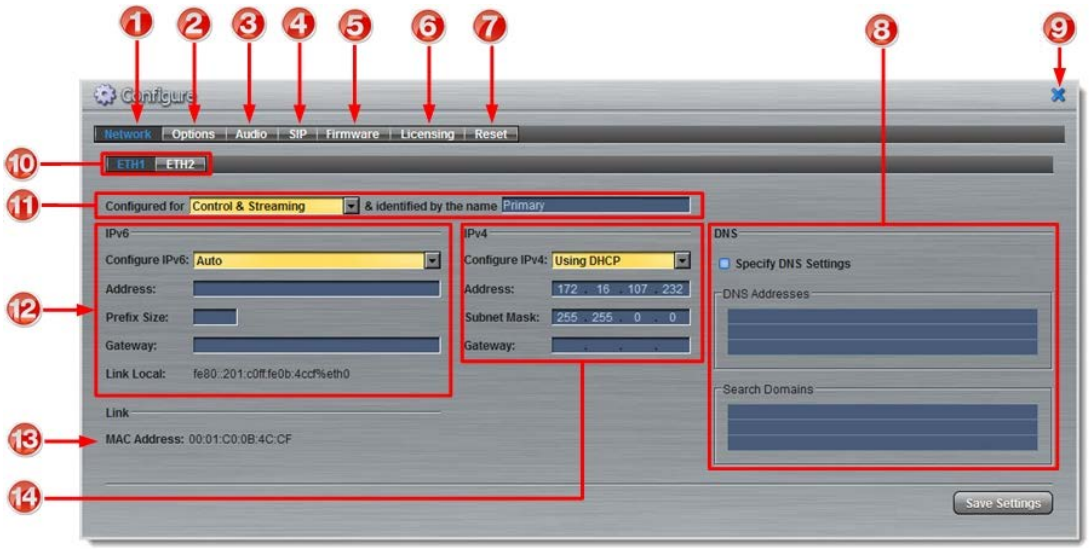
Configure Panel



	Feature	Description
1	**Network tab**	Click to edit or view codec network configuration settings
2	**Options tab**	Click to configure RS232 and QoS data settings.
3	**Audio tab**	Click to configure the AES Output Clock sample rate
4	**SIP tab**	Click to edit or view SIP configuration settings
5	**Firmware tab**	Click to view software versions and perform an upgrade
6	**Licensing tab**	Click to select a license file and install it into the codec
7	**Reset tab**	Click to reset codec default settings
8	**DNS** Pane	Activate to specify DNS addresses and domains to search.
9	**Close** button	Click to close the **Configure** panel
10	Network Interface	Select a network interface for configuration options
11	Network Interface Identifier	Control and streaming configuration options for each network interface, e.g. Ethernet Port 1 or 2.
12	**IPv6** details	IPv6 addressing details and configuration
13	**MAC Address**	Device MAC address
14	**IPv4** details	IPv4 addressing details and configuration

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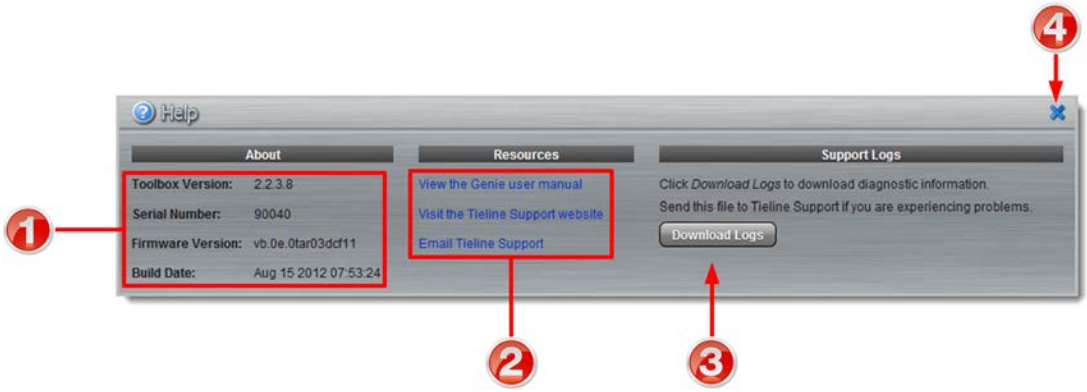
Help Panel



	Feature	Description
1	About	Details of the Toolbox web-GUI and codec firmware versions, as well as the codec serial number
2	Resources	Links to open the user manual in a new browser, or view support information
3	Support Logs	Click to download diagnostic information that can be sent to Tieline support
4	**Close** button	Click to close the **Help panel**

Language Selection

The Toolbox web-GUI offers language support for several languages.

1. Click on the language drop-down menu arrow in the top right-hand corner of the web-GUI page.
2. Select your language of choice.

3. Click to refresh your web-browser and display the new language selected.

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14.1 Configuring IP Settings

IP connections can be configured using the **Network** panel in the web-GUI.

Important Note: For assistance with configuration of IPv4 or IPv6 network connections contact your IT Administrator.

IPv4 versus IPv6

An IP address is an unique address to identify a device on a TCP/IP network. Your codec uses dual IP protocol stacks to allow your codec to work on both IPv4 and IPv6 networks. Your Tieline codec supports both DHCP (default) IP addressing and static IP addresses for dialing IPv4 connection end-points.

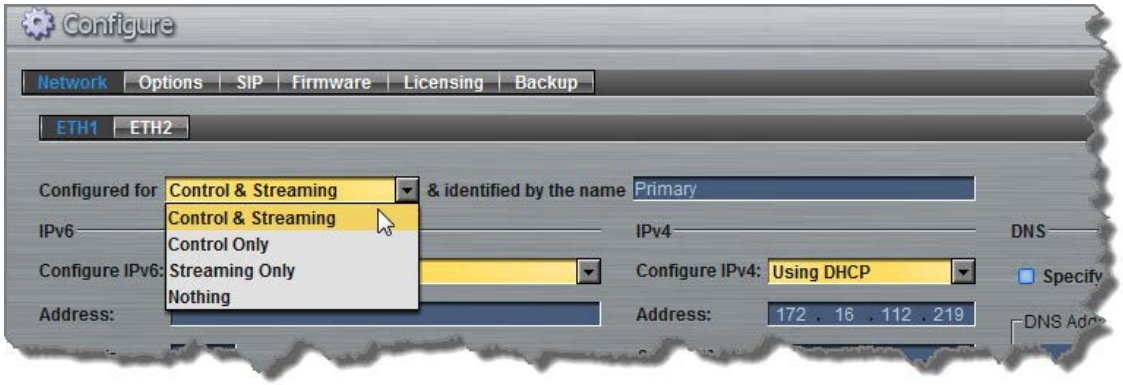
If you want to dial a codec with a public IP address you simply dial the IP address to connect. If you want to dial a codec with a private IP address you need to perform network address translation (NAT). NAT allows a single device, such as a broadband router, to act as an agent between the public internet and a local private LAN. Usually this will be set up at the studio end so you can dial into the studio from the remote codec.

Support for IPv6 connections allows you to use IPv6 infrastructure to connect to other codecs globally.

Configuring Ethernet Ports

Dual Ethernet ports allow two different IP network connections. These connections can be configured for:

- Streaming audio: stream audio only from an Ethernet port.
- Controlling audio: codec control and command only from the Ethernet port.
- Controlling and Streaming: stream audio and control and command the codec via the Ethernet port.
- Nothing: Disable the Ethernet port from streaming audio and codec command and control.



The name entered into the right-hand text box, e.g. **Primary** or **Secondary**, is an identifier used when configuring new programs via the **Programs panel**.

IPv4 Address Configuration

The codec is capable of automatic DHCP address assignment, or manually programmed static IPv4 address configuration via the drop-down **Configure IPv4** menu. If you want to ignore IPv4 settings select **Off**.

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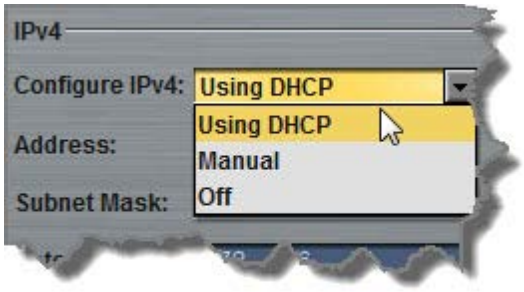
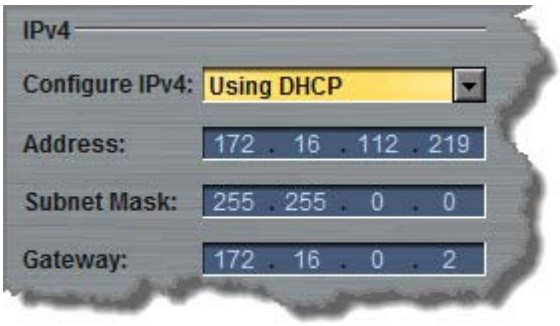
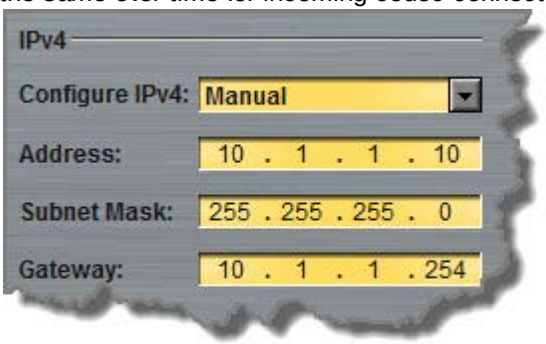


DHCP IP addresses are automatically assigned and can change each time you connect to your Internet Service Provider or to your own local area network (LAN). By default the codec is programmed for DHCP-assigned IP addresses.



Static IP addresses are fixed addresses that are recommended for studio installations, so that IP address dialing remains the same over time for incoming codec connections.



Click **Save Settings** to store all configuration settings.

Note: The **Subnet Mask** is used by the TCP/IP protocol to determine whether a host is on the local subnet or on a remote network. The default **Gateway** is the router linking the codec's subnet to other networks. See your IT administrator for more details.

IPv6 Address Configuration

An IPv6 address is represented by 8 groups of 16-bit hexadecimal values separated by colons (:). The drop-down **Configure IPv6** menu provides three address configuration options:

1. Auto: An address is automatically assigned to the codec when you connect the codec to an IPv6 router. This process is similar to how an IPv4 DHCP address is assigned.
2. Manual: Select to manually enter IPv6 address details.
3. Off: Select to ignore IPv6 address details.

Important Note: Select **Off** in the drop-down **Configure IPv6** menu if you are not using

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IPv6 to connect to another device. This ensures your codec will attempt to connect using IPv4 at all times.

Types of IPv6 Addresses

There are two types of addresses displayed in the IPv6 section:

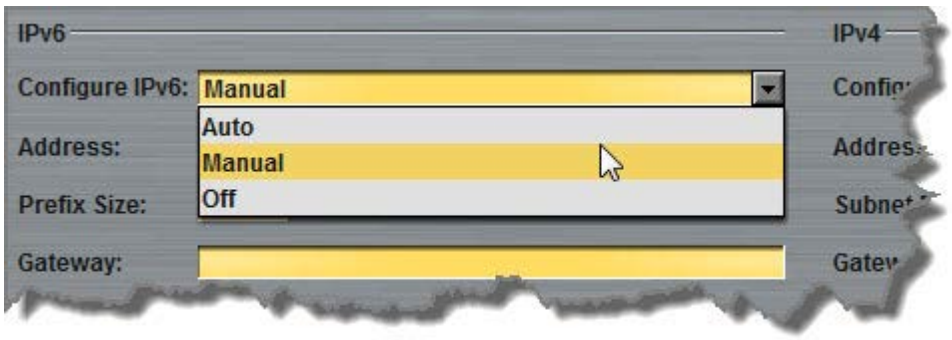
1. IPv6 address (normally global): A router-allocated IP address with 'global' visibility, details of which are displayed in the **Address, Prefix** and **Gateway** text boxes.
2. Link Local Address: A local address which can only be used to connect to another device directly over a LAN. This address is allocated by the codec internally based on MAC address details.

Auto Address Assignment

By default the codec is programmed for connecting the codec to an IPv6 router which automatically allocates IPv6 address details, as displayed in the following example.

Manual IPv6 Address Assignment

To program IPv6 address details into the codec manually, select **Manual** and enter details into the **Address, Prefix** and **Gateway** text boxes.



Click **Save Settings** to store all configuration settings.

Specifying DNS Settings

It is possible to specify Domain Name Server (DNS) settings to allow easy look up of codecs within the specified **DNS Addresses** or **Domains**.

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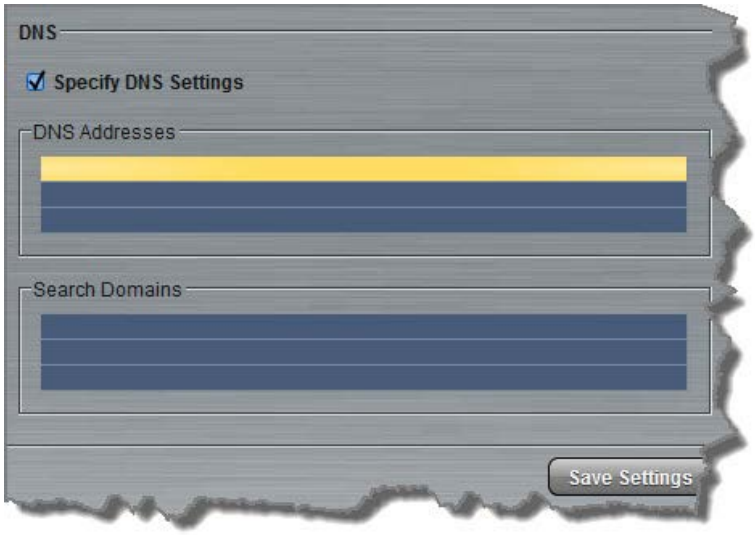

Configure QoS

1. Open the web-GUI and click the **CONFIGURE** button at the top of the screen to display the **Configure panel**.
2. Click the **Options** button.
3. Click in the **DSCP** field and enter the priority setting recommended by your IT administrator.
4. Click **Save settings**.

14.2 Configuring Input Settings

Click the **Inputs symbol** to view input controls available within the Toolbox web-GUI.

Important Note: 15 volt phantom power can only be supplied on the Auxiliary input; this is disabled by default.

Configuring Input Channel Settings

Click the **Configure** symbol on each input to adjust input controls.

Renaming Input Channels:

1. Click the **Configure** symbol on the input channel you want to rename.
2. Select **Name** and click in the text box to edit or enter a new name.
3. Click **Change Name** to confirm the name change.

Selecting Analog and Digital Audio Sources:

Codec inputs are configured for analog high-gain mic level audio sources by default.

1. Click the **Configure** symbol on channel 1.

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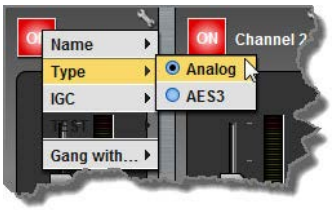
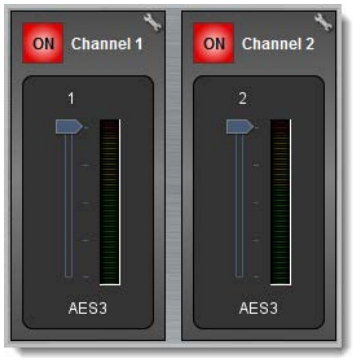
2. Select **Type** and click to select either **Analog** or **AES3**.



3. When you select AES3, the display changes to reflect 100% input levels; slider and input on/off controls are locked on.

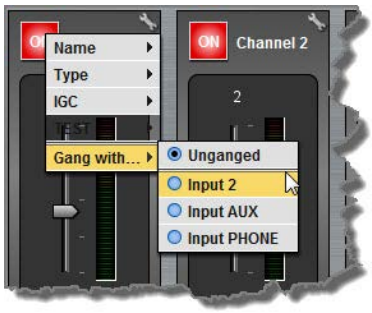


Important Note: Input levels can only be adjusted on analog inputs. Digital AES3 source audio is not adjustable. See Configuring AES3 audio for more information about the digital inputs and outputs.

Ganging Channels:

Ganging is useful because it allows you to adjust the audio level of both inputs simultaneously.

1. Click the **Configure** symbol on either channel.
2. Select **Gang** and click to either gang or ungang channels.



3. When ganged, the two channel sliders move in sync with each other when dragged using a mouse-pointer.

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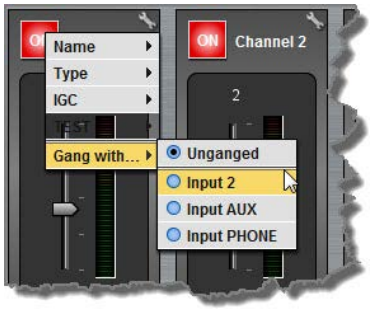
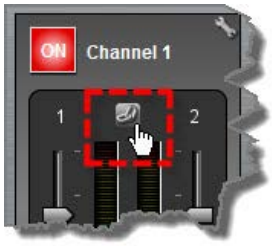


4. Click the **Link** symbol to temporarily disable the ganging function and fine-tune channel audio levels. Click the **Link** symbol again to resume ganging.

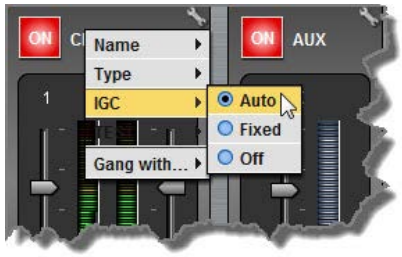


Setting Audio Levels

Audio levels on the **Input panel** should be set to ensure audio peaks average at the first yellow indications on the PPM meters, which represents +4dBu. These levels should also be checked against the **Input PPM Meters** on the **Master panel**.

Other Input Controls

Adjust the **IGC** (Intelligent Gain Control) input settings to **Auto**, **Fixed** or **Off** as required.



Important Note: When the auxiliary input (**AUX IN**) is **On** the default mixer configuration sends audio to inputs 1 and 2 for two channel program configurations and to input 1 for a mono program. If you are not using the Aux input ensure it is **Off** to avoid additional noise in program audio.

Locking Input Settings

1. Click the **Lock** symbol to lock all **Input panel** settings.
2. When locked, the **Input panel** is greyed out and the lock symbol appears in the bottom-left corner.

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14.3 Configuring Merlin Point-to-Point Programs

Important Notes: Before you start program configuration please note:

- You cannot edit a program when it is currently loaded in the codec.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- To learn more about programs see the section titled About Program Dialing.

The **Programs panel** incorporates a wizard to configure your new program and all audio stream settings. Before you configure a new codec program consider if:

- You want your codec to be capable of dialing and answering, dialing only or answering only.
- A backup connection is required.

This section contains instructions for:

1. Configuring Merlin Point-to-Point Programs: Dialing
2. Configuring a Merlin Backup Connection or Auto Reconnect
3. Configuring Merlin to Answer Connections

For more information about programs and audio streams within programs see the section titled About Program Dialing.

Note: The following connection setup instructions will display how to configure a dial and answer program, with a backup connection. If you want the codec to either dial or answer only, select the option and the wizard will automatically display relevant screens to allow you to configure the codec correctly.

Configuring Merlin Point-to-Point Programs: Dialing

1. Open the web-GUI and click the **Programs** button at the top of the screen to display the **Programs panel**.

Programs
New Program...
STL Studio 1 NYC
To create a program, click New Program...
A Program configures a Tieline codec to send or receive one or more Audio Streams based upon the particular application the codec is being used for at any given time.
Each Audio Stream within a Program can be defined separately and contain a variety of settings relating to the number of connections (e.g. primary and backup) and the number of destinations to which each Audio Stream is distributed.
More information?

2. Click the **New Program** button to open the wizard and:
   - Click in the **Program Name** text box to name the new program.
   - Select **Peer to Peer,** or if you want to use an existing program as a template, select this option. Then click **Next**.

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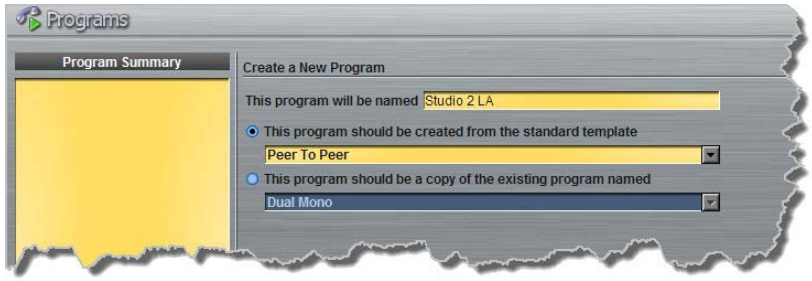
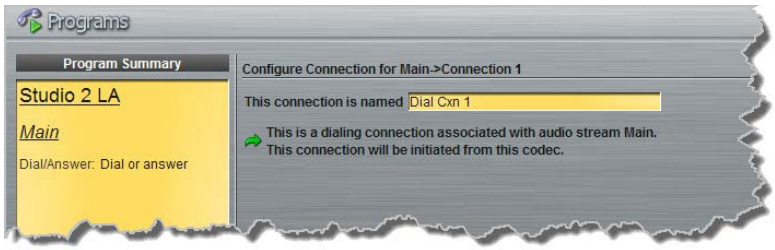

Programs
Program Summary
Create a New Program
This program will be named Studio 2 LA
This program should be created from the standard template
Peer To Peer
This program should be a copy of the existing program named
Dual Mono

Important Notes: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

3. Enter a name for the **Audio Stream** and configure the codec to dial, answer or dial and answer. Then click **Next**.

Programs
Program Summary
Studio 2 LA
Configure Audio Stream 1 for Studio 2 LA
This audio stream is named Main
The connections associated with this audio stream will allow this codec to:
Dial or answer
Dial only
Answer only

4. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.


Programs
Program Summary
Studio 2 LA
Main
Dial/Answer: Dial or answer
Configure Connection for Main->Connection 1
This connection is named Dial Cxn 1
This is a dialing connection associated with audio stream Main.
This connection will be initiated from this codec.

5. Follow the instructions on the right-hand side of the panel to configure the transport settings for the connection, then click **Next**.

Programs
Program Summary
Studio 2 LA
Main
Dial/Answer: Dial or answer
Main->Dial Cxn 1
Direction: Dialing
Configure Transport settings for Main->Dial Cxn 1
Transport: IP
Audio Protocol: IDP/IP
Session Protocol: Tieline Codecs
Encode/Decode: Both
Connect Timeout: 2 min
Enable Auxiliary Data
1. Select the transport type to be used for this connection.
2. Select the audio streaming protocol to be used for this connection.
3. Select the session protocol type to be used for this connection.
4. Indicate whether this connection will encode only, decode only, or encode & decode audio.
5. Specify the duration after which a connection attempt will time out.
6. Select the Enable Auxiliary Data check-box to transmit auxiliary data with the encoded audio data.
< Back
Next >
Save Program
Cancel

Important Note: Auxiliary IP data connections are not possible between the codec and G3 Commander and i-Mix codecs. Ensure this setting is disabled when connecting with these codecs, or connection attempts will fail. See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

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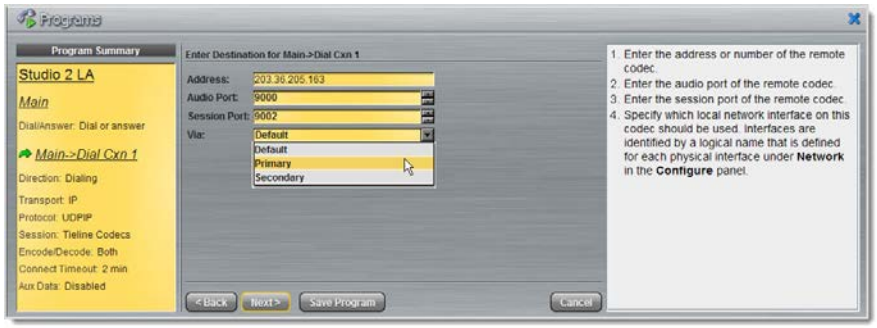
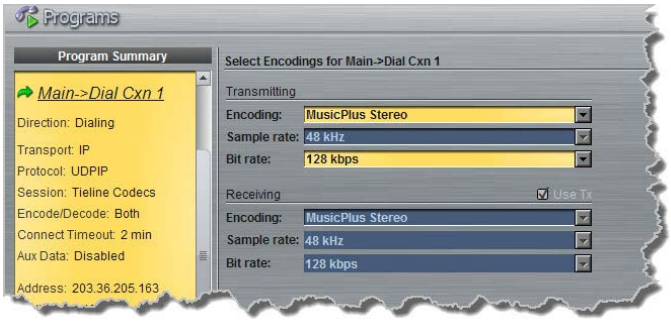
6. Configure destination codec details and ports, then specify which streaming interface is used to dial this connection, e.g. **Primary** (Ethernet port 1) or **Secondary** (Ethernet port 2).

At this point you can click **Save Program** and save the program with the last algorithm, jitter and FEC settings physically entered into the codec without backup. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings for this connection and configure backup audio for this audio stream (recommended).

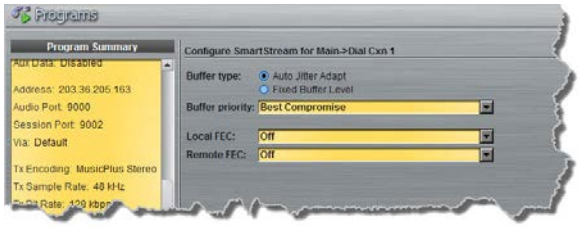

Programs
Program Summary
Studio 2 LA
Main
Dial/Answer: Dial or answer
Main->Dial Cxn 1
Direction: Dialing
Transport: IP
Protocol: UDPIP
Session: Tieline Codecs
Encode/Decode: Both
Connect Timeout: 2 min
Aux Data: Disabled
Enter Destination for Main->Dial Cxn 1
Address: 203.36.205.163
Audio Port: 9000
Session Port: 9002
Via: Default
Default
Primary
Secondary
1. Enter the address or number of the remote codec.
2. Enter the audio port of the remote codec.
3. Enter the session port of the remote codec.
4. Specify which local network interface on this codec should be used. Interfaces are identified by a logical name that is defined for each physical interface under Network in the Configure panel.
< Back
Next >
Save Program
Cancel

7. Click the drop-down arrows on the right-hand side of each text box to select the **Encoding**, **Sample rate**, **Bit-rate** or **Sample size** options. Click **Next** to continue.



8. Click to configure:
   - **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer Priority**, or
   - **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 1000ms depending on the algorithm you select.
   - **Local** and **Remote FEC** settings if required.

Click **Save Program** to save all settings, or click **Next** to configure **Auto Reconnect** or a backup connection using fail over.



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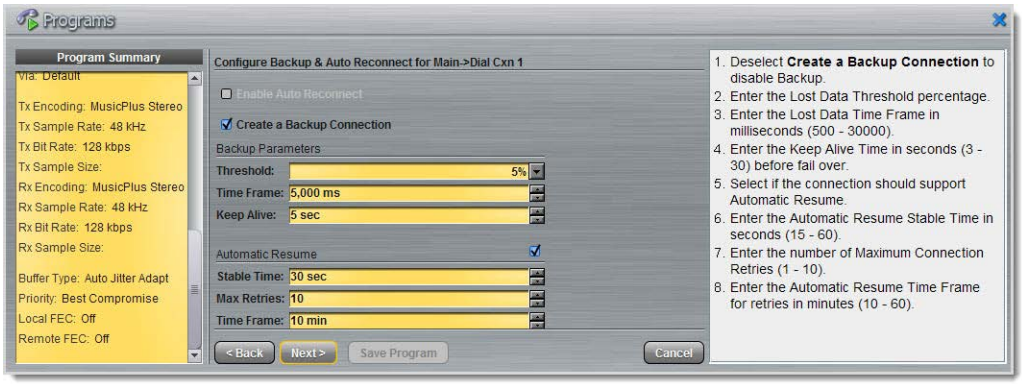
Configuring a Merlin Backup Connection or Auto Reconnect

At this point in the wizard you can choose to configure **Auto Reconnect** or create a backup connection for the audio stream you are configuring.

Important Note: When **Auto Reconnect** is enabled, the dialing codec will continue to attempt a connection with the remote codec until **Disconnect** is pressed either on the dialing codec's keypad, or in the web-GUI.

To configure a backup connection:

1. Click to select the check-box for **Create a Backup Connection**. Adjust the parameters and click **Next**.


Programs
Program Summary
Via: Default
Tx Encoding: MusicPlus Stereo
Tx Sample Rate: 48 kHz
Tx Bit Rate: 128 kbps
Tx Sample Size:
Rx Encoding: MusicPlus Stereo
Rx Sample Rate: 48 kHz
Rx Bit Rate: 128 kbps
Rx Sample Size:
Buffer Type: Auto Jitter Adapt
Priority: Best Compromise
Local FEC: Off
Remote FEC: Off
Configure Backup & Auto Reconnect for Main->Dial Cxn 1
Enable Auto Reconnect
Create a Backup Connection
Backup Parameters
Threshold: 5%
Time Frame: 5,000 ms
Keep Alive: 5 sec
Automatic Resume
Stable Time: 30 sec
Max Retries: 10
Time Frame: 10 min
< Back
Next >
Save Program
Cancel
1. Deselect Create a Backup Connection to disable Backup.
2. Enter the Lost Data Threshold percentage.
3. Enter the Lost Data Time Frame in milliseconds (500 - 30000).
4. Enter the Keep Alive Time in seconds (3 - 30) before fail over.
5. Select if the connection should support Automatic Resume.
6. Enter the Automatic Resume Stable Time in seconds (15 - 60).
7. Enter the number of Maximum Connection Retries (1 - 10).
8. Enter the Automatic Resume Time Frame for retries in minutes (10 - 60).

Note: The explanations within the following table can be used to assist with backup connection configuration.

	Screen Display	Description
1	Threshold	The percentage of lost data measured during a given time frame
2	Time Frame	The time frame against which lost data is measured
3	Keep Alive	The keep connection alive time before failing over to a backup connection; Tieline RTP pings every second to confirm connectivity
4	Automatic Resume	Select the check-box to configure fail back to a higher priority connection
5	Stable Time	The amount of time a primary connection must remain stable before attempting to fail back from the backup connection
6	Maximum Retries	The maximum number of fail back retries a codec can try before ending fail back attempts
7	Time Frame	The time frame used to measure the number of fail back retries attempted

2. Enter a name for the backup connection and click **Next**.

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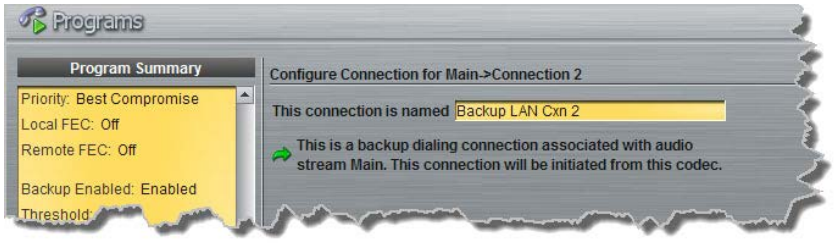
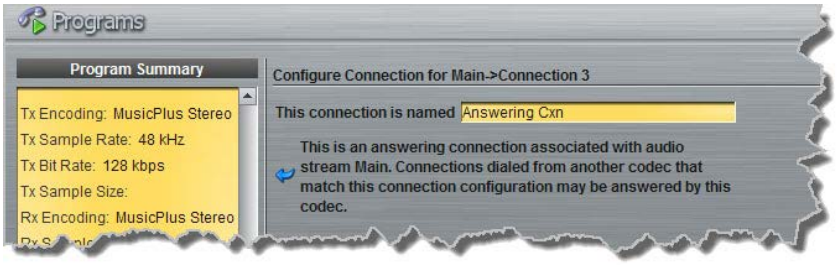

Programs
Program Summary
Priority: Best Compromise
Local FEC: Off
Remote FEC: Off
Backup Enabled: Enabled
Threshold:
Configure Connection for Main->Connection 2
This connection is named Backup LAN Cxn 2
This is a backup dialing connection associated with audio stream Main. This connection will be initiated from this codec.

3. Click **Next** to continue through the wizard and configure the backup connection in a similar manner to how you configured the primary connection.

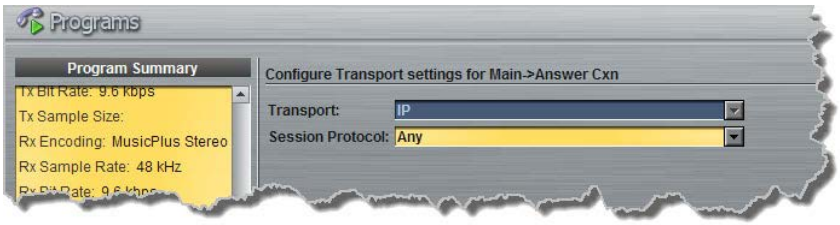
Configuring Merlin to Answer Connections

If you are configuring the codec to allow it to answer one or more incoming audio stream connections:

1. Enter a name for the answering connection and click **Next**.


Programs
Program Summary
Tx Encoding: MusicPlus Stereo
Tx Sample Rate: 48 kHz
Tx Bit Rate: 128 kbps
Tx Sample Size:
Rx Encoding: MusicPlus Stereo
Configure Connection for Main->Connection 3
This connection is named Answering Cxn
This is an answering connection associated with audio stream Main. Connections dialed from another codec that match this connection configuration may be answered by this codec.

2. Configure the transport settings and click **Next**.



3. Configure jitter buffer and FEC settings. You then have 2 options:

    i. If you want to create another answering connection, select the check-box for **Create another answering connection** and continue through the wizard.
    ii. Click **Save Program** to save the program at this point.

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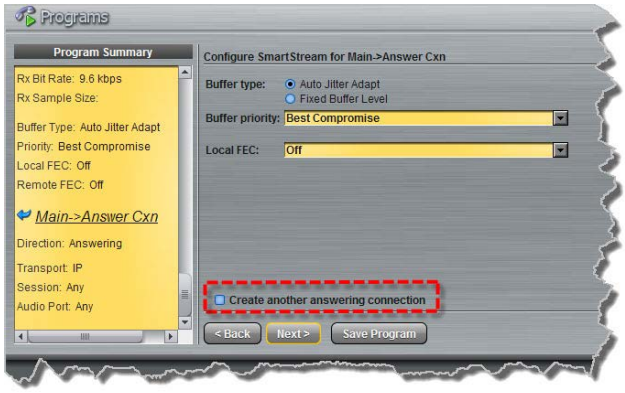

4. Select the check-box if you want to connect the program immediately, then click **Finish**.

Programs
You've successfully created a program. You can connect this program from the Master panel.
Connect this program when I click Finish
< Back
Next >
Finish
Close

5. The newly created program will be displayed in the **Programs panel** and in the **Master panel**.

Programs
New Program...
Studio 2 LA
Main
Dial Cxn 1
Backup LAN Cxn 2
Answer Cxn 3
Studio 2 LA
Edit Name · Delete Program
Dial Cxn 1 (Main) Direction: Dialing Transport: IP Address: 203.36.205.163
Backup LAN Cxn 2 (Main) Direction: Dialing Transport: IP Address: 203.36.205.163
Answer Cxn 3 (Main) Direction: Answering Transport: IP

Dial the program by loading and connecting using the **Master panel**, or dial the program manually using the codec front panel.

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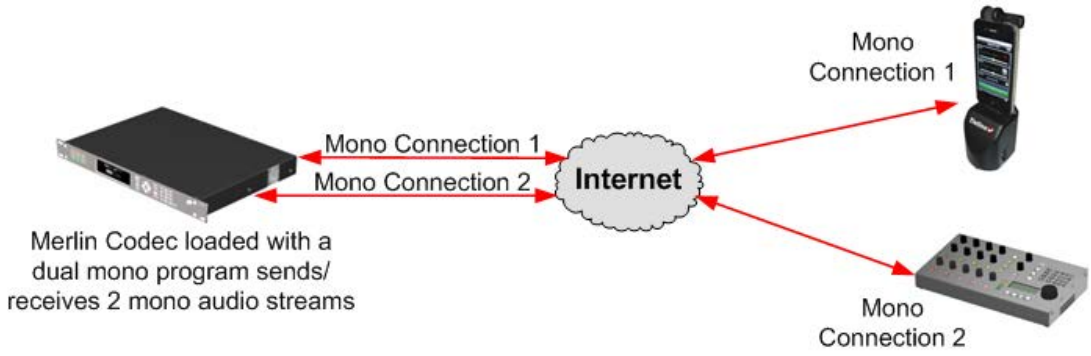
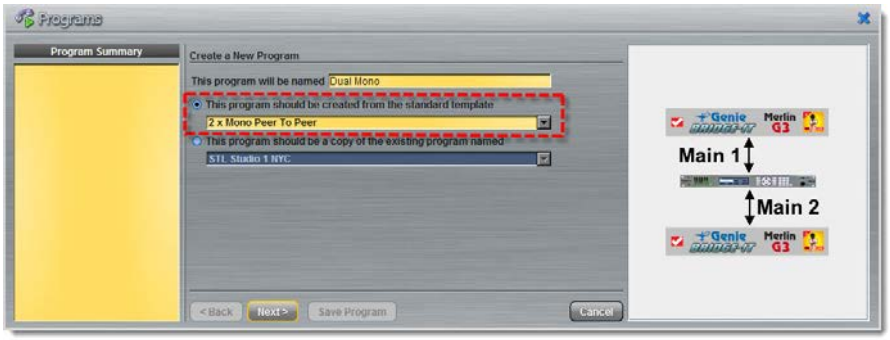
14.4 Configuring Dual Mono Programs

A dual mono program allows a codec to create two simultaneous audio stream connections with different destination codecs.



Configuring a program to dial two mono audio stream connections from a codec is very similar to creating a single audio stream connection. The main difference is the selection of **2 x Mono Peer to Peer** option from the drop-down menu in the first screen of the program wizard.


Programs
Program Summary
Create a New Program
This program will be named Dual Mono
This program should be created from the standard template
2 x Mono Peer To Peer
This program should be a copy of the existing program named
STL Studio 1 NYC
< Back
Next >
Save Program
Cancel
Main 1
Main 2

The program wizard will then progress through the various configuration screens and configure **Audio Stream 1** and then **Audio Stream 2** in the same way as a program configured with a single audio stream.

See Configuring Merlin Point-to-Point Programs for more details about individual settings within the program wizard.

Routing Incoming Dual Mono Audio Streams to Specific Codec Outputs

If your intention is to ensure incoming dual mono audio stream connections are always routed to the same outputs configure a new answering program as follows:

1. Open the web-GUI and click the **Programs** button at the top of the screen to display the **Programs panel**.
2. Click the **New Program** button to open the wizard and:
   - Click in the **Program Name** text box to name the new program.
   - Select **2 x Mono Peer to Peer,** or if you want to use an existing program as a template, select this option. Then click **Next**.
3. Enter a name for the **Audio Stream** and select **Answer only**. Then click **Next**.

Note: if the codec you are configuring needs to dial connections as well as answer, you can select **Dial or answer** *and configure both dialing and answering connections. For the purposes of this example dialing is excluded.*

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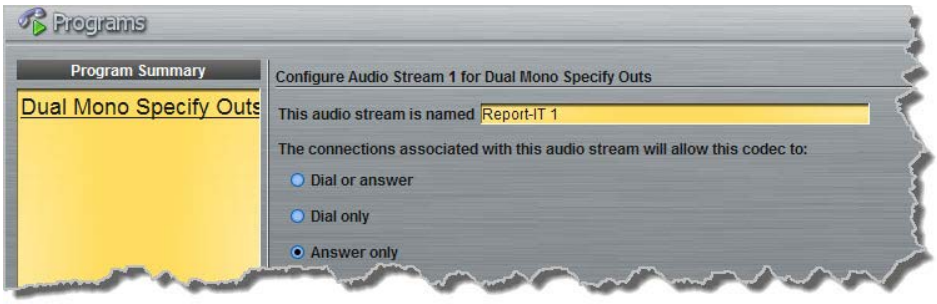
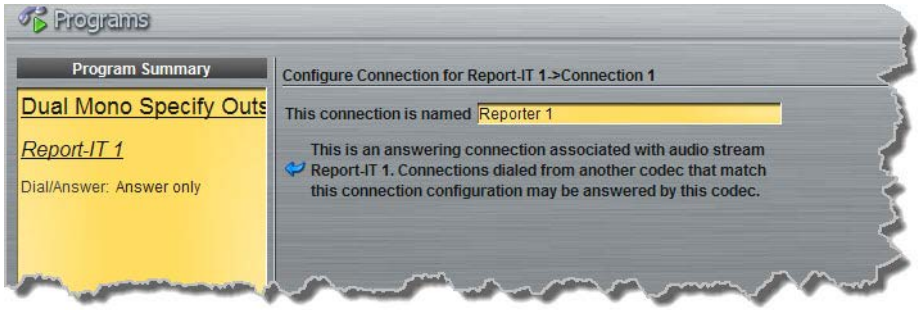
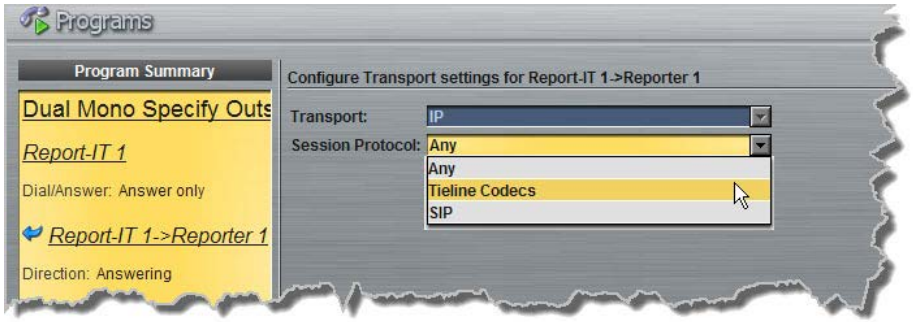

Programs
Program Summary
Dual Mono Specify Outs
Configure Audio Stream 1 for Dual Mono Specify Outs
This audio stream is named Report-IT 1
The connections associated with this audio stream will allow this codec to:
Dial or answer
Dial only
Answer only

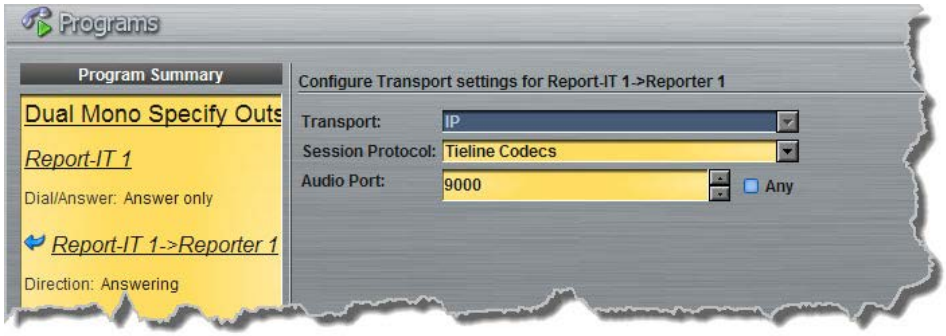
4. Enter the name of the connection in the text box, then click **Next**.


Programs
Program Summary
Dual Mono Specify Outs
Report-IT 1
Dial/Answer: Answer only
Configure Connection for Report-IT 1->Connection 1
This connection is named Reporter 1
This is an answering connection associated with audio stream Report-IT 1. Connections dialed from another codec that match this connection configuration may be answered by this codec.

5. Click the drop-down **Session Protocol** menu and select **Tieline Codecs**.



6. Click to deselect the **Any** check-box.



Important Notes: This ensures that any codec dialing and using the port specified will always be routed to codec output 1. By default port 9000 is the port specified for the first IP connection made by any Tieline codec; port 9010 is used by default for a second IP connection. If you need to change this setting on your dialing codec, or in your Report-IT smart phone application, this will need to be adjusted accordingly in the answering codec.

7. Click to configure the jitter buffer and FEC settings if required.

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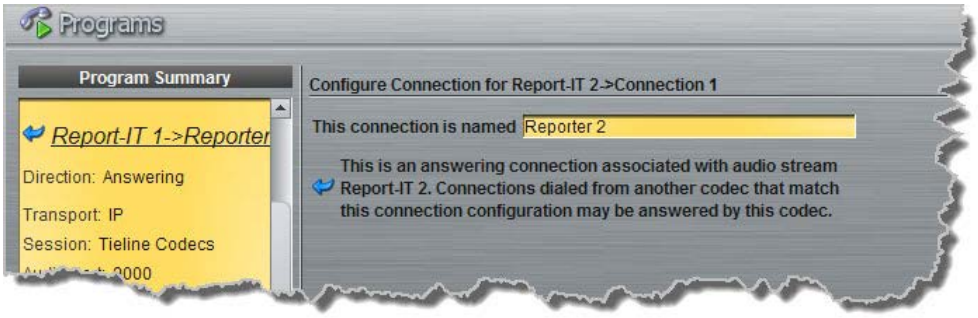
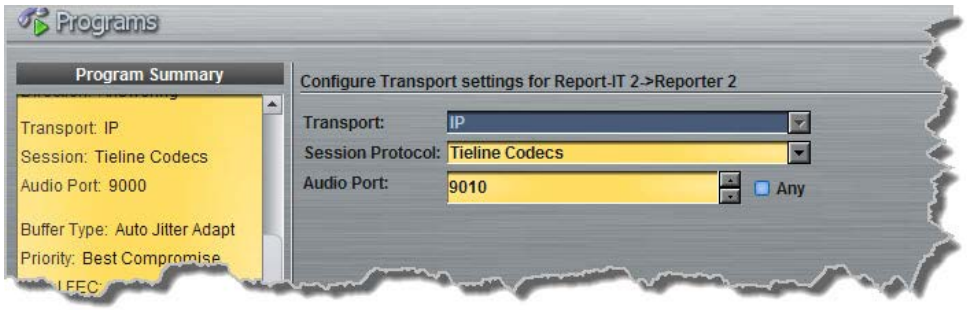
8. Enter a name for the second **Audio Stream** and select **Answer only**. Then click **Next**.
9. Enter the name of the second audio stream connection in the text box and click **Next**.


Programs
Program Summary
Report-IT 1->Reporter
Direction: Answering
Transport: IP
Session: Tieline Codecs
Configure Connection for Report-IT 2->Connection 1
This connection is named Reporter 2
This is an answering connection associated with audio stream Report-IT 2. Connections dialed from another codec that match this connection configuration may be answered by this codec.

10. Click the drop-down **Session Protocol** menu and select **Tieline Codecs**.
11. Click to deselect the **Any** check-box and then click **Next**.



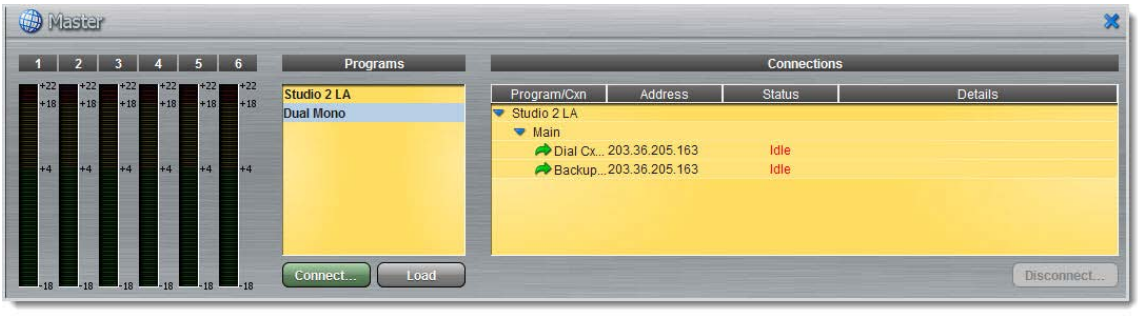
12. Click to configure the jitter buffer and FEC settings if required.
13. Click **Save Program** to save settings.

If the previous configuration is used then any codec dialing over IP1 (using default Tieline IP port settings) will be routed to output 1 on the codec and IP2 will be routed to output 2.

14.5 Dial/Disconnect Multiple IP Streams

Load and Connect Multiple Audio Streams within a Program

1. Click to select the program you want to load from the **Programs** list.



2. Click **Connect** to load the program and connect all audio streams.

Disconnect All Audio Stream Connections

1. Click to select the program in the **Connections** pane, e.g. "Dual Mono" in the following example.

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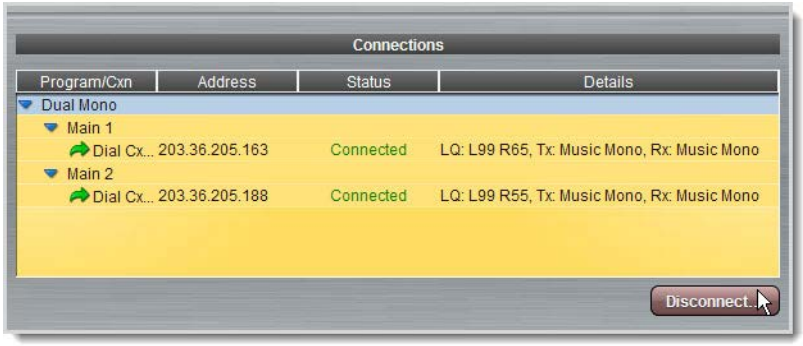
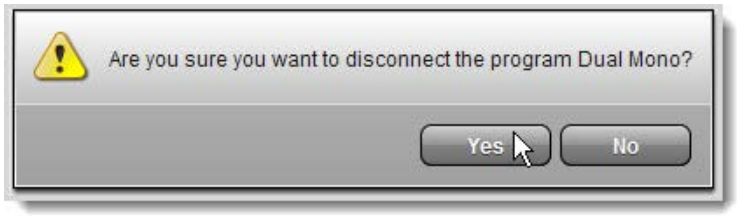


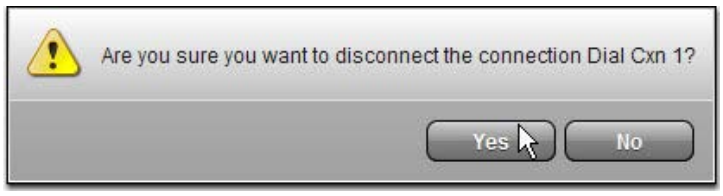
2. Click the **Disconnect** button.
3. Click **Yes** in the confirmation dialog to disconnect all audio stream connections.



Disconnect a Single Audio Stream Connection

1. Click to select the audio stream connection you want to disconnect.

2. Click the **Disconnect** button.
3. Click **Yes** in the confirmation dialog to disconnect all audio stream connections.



14.6 Viewing/Editing Program Settings

Important Notes: You cannot edit or delete a program when it is currently loaded in the codec; ensure you have unloaded a program prior to editing the current configuration.

To view configuration settings for an existing program, or edit settings:

1. Open the web-GUI and click the **Programs** button at the top of the screen to display the

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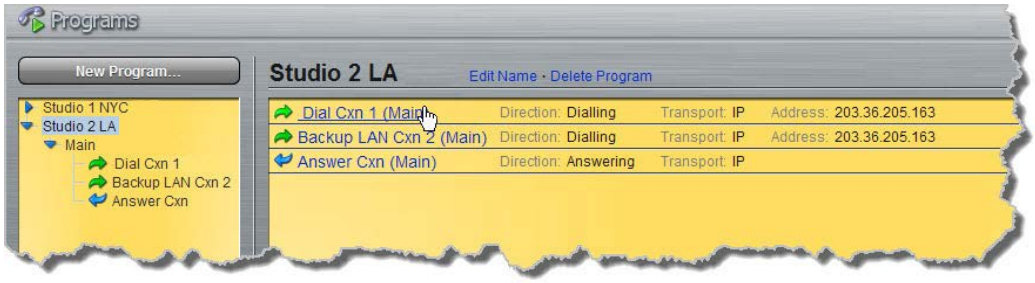
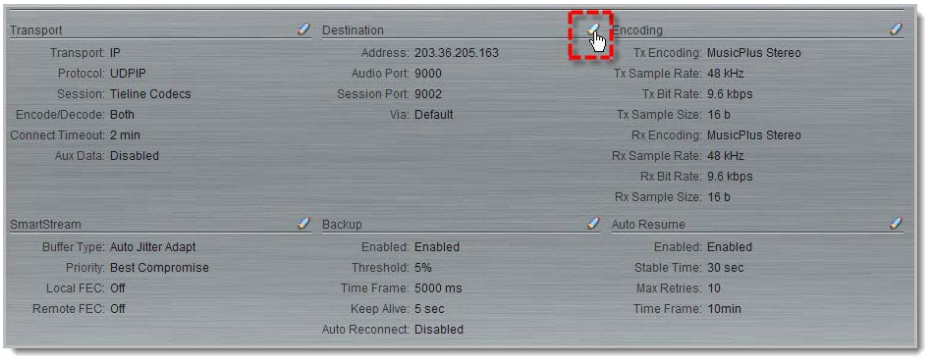
Programs panel.

2. Click the blue arrow next to a program to expand and display audio stream connections, then click the connection in the right hand pane which you want to view or edit.



3. Click the **Edit** symbol to adjust configuration settings displayed in the panel.



4. The program wizard will open at the relevant point to facilitate editing of connection parameters. Click **Save Program** to store settings.

Deleting Programs

1. Ensure the program you want to delete is not currently loaded in the codec.
2. Click to select a program listed on the left hand side of the Programs panel.
3. Click **Delete Program** next to the program name.

14.7 Configuring SIP Settings

The codec is fully EBU N/ACIP Tech 3326 compliant when connecting using SIP (Session Initiation Protocol) to other brands of IP codecs.

About SIP

SIP provides superior interoperability between different brands of codecs due to its standardized protocols for connecting devices and is intended to be used when connecting Tieline codecs to non-Tieline devices. Devices primarily use SIP to dial another device's SIP address and find its location with a minimum of fuss. This task is usually performed by SIP servers, which communicate between SIP-compliant devices to set up a call.

When connecting two devices, SDP performs similar tasks to Tieline's proprietary session data, which is used to configure all non-SIP IP connections. There are two very distinct parts to a call when dialing over IP. The initial stage is the call setup stage and this is what SIP is used for. The second stage is when data transference occurs and this is left to the other protocols used by a device (i.e. using UDP to send audio data).

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All the mandatory EBU N/ACIP 3326 algorithms are supported (G.711, G.722, MPEG-1 Layer 2 and 16 bit PCM), as well as optional algorithms including LC- AAC, HE-AAC and E-apt-X. The default algorithm selected when connecting using SIP is G.711.

Important Notes:

- Each codec should be registered to a different SIP server account to avoid connection conflicts.
- SIP dialing is only supported over point-to-point connections, not multi-unicast connections.
- Tieline G3 codecs do not support connections using AAC and will default to MPEG Layer 2 if an incoming call is programmed to use this algorithm.

SIP Server Connections: Getting Started

Registering codecs for SIP connectivity is simple. First, choose the SIP server that you wish to register your codec with. On a LAN this may be your own server, or it could be one of the many internet servers available. We recommend that you use your own SIP server and configure it to use G.711, G.722, MP2 and AAC algorithms. This is because most internet SIP servers are for VoIP phones and are only configured for G.711 and GSM algorithms.

When you register an account with a SIP server you will be provided with:

- The SIP server IP address.
- A username (often the same as a SIP number).
- A password.
- Domain details.
- Realm details (sometimes).

Program the Codec for SIP using the Web-GUI

Use the Toolbox web-GUI to program SIP account registration details into your codec. Once these details have been entered into the codec, each time it is connected to a public IP address it will contact the SIP server automatically to acknowledge its presence over a wide area network.

1. Connect your codec to a LAN connection with a public IP address, then login to the Toolbox web-GUI and click the **Configure** button at the top of the screen to display the **Configure panel**.
2. Click the **SIP** button.
3. Enter the account details into the relevant text boxes.
4. Click to select **Activate Account** and click the **Save Settings** button to create the account in the codec. A confirmation message is displayed in the bottom-left corner of the **Settings panel** if the account details are saved successfully.

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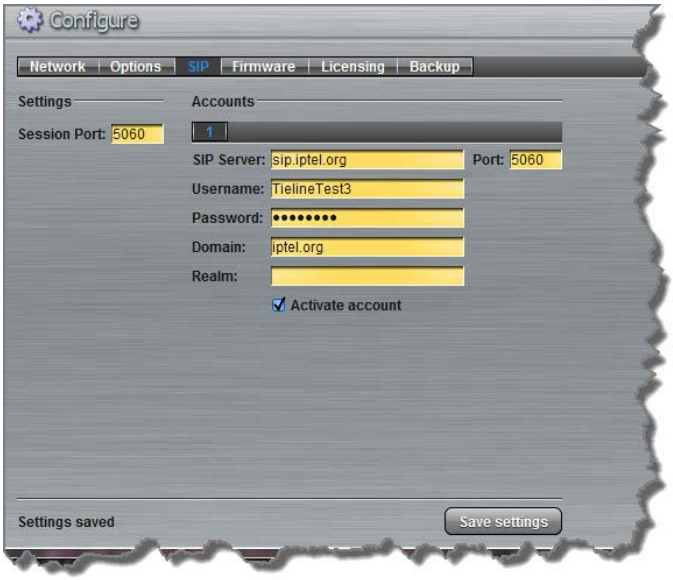

5. Enable SIP within the codec via the **CONFIGURE** button, then navigate to **SIP > Accounts > Select Account name > Active [Enabled].** After selecting **Enabled**, press the **RETURN** button to navigate backwards and make sure that the codec has been registered to the SIP server account by checking the registration symbol appears as per the following example.

Important Notes: Some 3G service providers may block SIP traffic over UDP port 5060. It is possible to reprogram this but we recommend that you contact our support desk at support@tieline.com before attempting this.

14.8 Configuring SIP Programs

SIP programs are like a normal IP program to configure, with two small differences; entering a SIP address and selecting SIP as the **Session Protocol**.

Important Notes: Before you start program configuration please note:
- SIP can only operate using port **ETH1** on the rear panel of the codec.
- You cannot edit a program when it is currently loaded in the codec.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- To learn more about programs see the section titled About Program Dialing.

1. Open the web-GUI and click the **Programs** button at the top of the screen to display the **Programs panel**.

2. Click the **New Program** button to open the wizard and:
   - Click in the **Program Name** text box to name the new program.
   - Select **Peer to Peer,** or if you want to use an existing program as a template, select this option. Then click **Next**.

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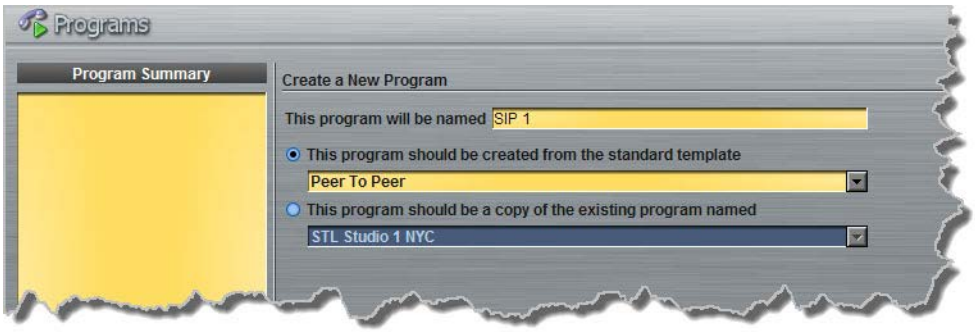
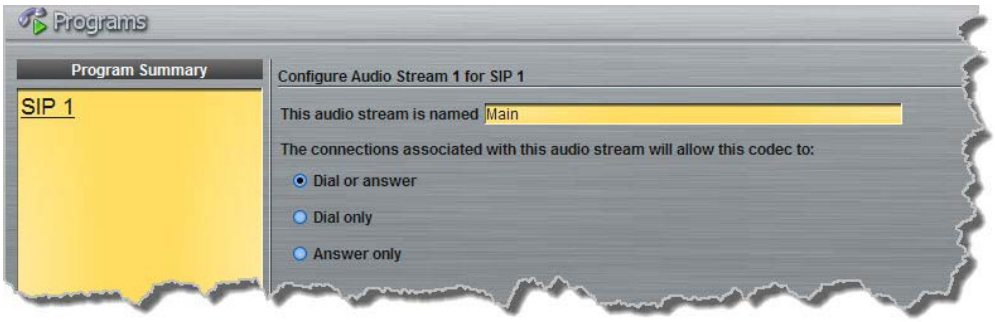

Programs
Program Summary
Create a New Program
This program will be named SIP 1
This program should be created from the standard template
Peer To Peer
This program should be a copy of the existing program named
STL Studio 1 NYC

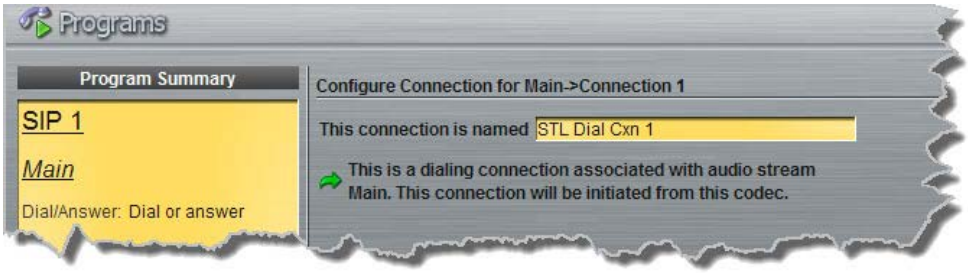
Important Notes: When you choose to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

3. Enter a name for the **Audio Stream** and configure the codec to dial, answer or dial and answer. Then click **Next**.

Note: The following example will display how to configure a dial and answer program. If you want the codec to either dial or answer only, select the option and the wizard will automatically display screens to allow you to configure the codec correctly.


Programs
Program Summary
SIP 1
Configure Audio Stream 1 for SIP 1
This audio stream is named Main
The connections associated with this audio stream will allow this codec to:
Dial or answer
Dial only
Answer only

4. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.


Programs
Program Summary
SIP 1
Main
Dial/Answer: Dial or answer
Configure Connection for Main->Connection 1
This connection is named STL Dial Cxn 1
This is a dialing connection associated with audio stream Main. This connection will be initiated from this codec.

5. Follow the instructions on the right-hand side of the panel to configure the transport settings for the connection: Ensure that you select:

- **IP** as the **Transport**.
- **UDPIP** as the **Audio Protocol**.
- **SIP** from the **Session Protocol** menu option.

Then click **Next**.

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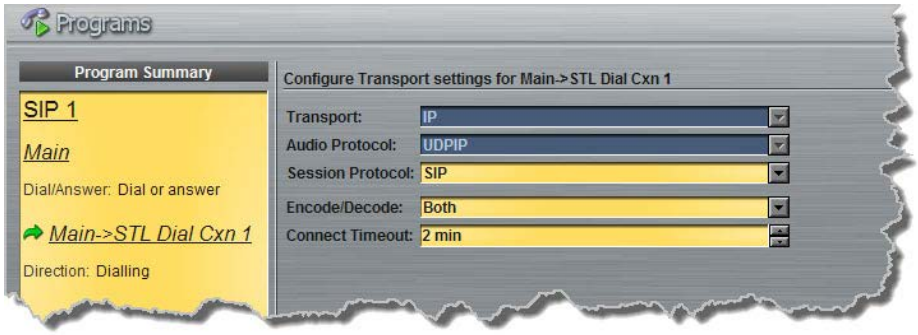
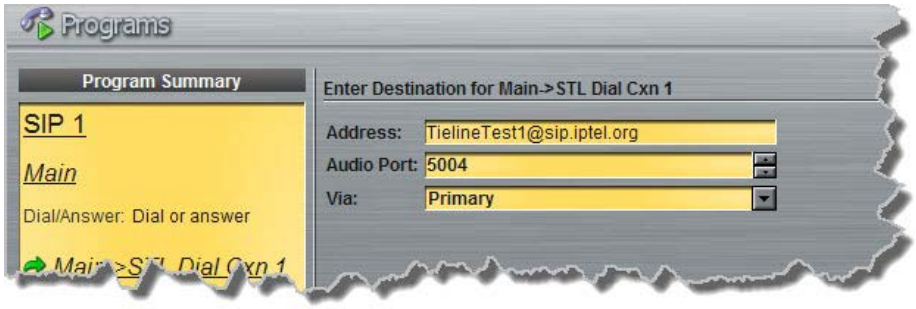


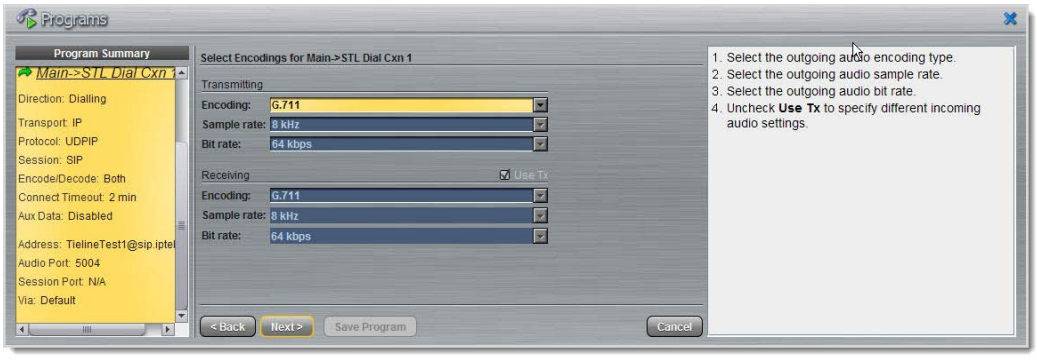
6. Configure the destination codec **Address** and **Audio Port**, then specify **ETH1** as the network interface used to dial the connection, e.g. **Primary** (Ethernet port 1).

At this point you can click **Save Program** and save the program with the last algorithm, jitter and FEC settings physically entered into the codec without backup. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings for this connection and configure backup audio for this audio stream (recommended).



Important Notes: The default UDP audio port when using SIP is 5004 in Tieline codecs. To contact a codec that is behind a firewall or NAT-enabled router, it is essential that this and all other relevant ports are open and forwarded to the other device.

7. Click the drop-down arrows on the right-hand side of each active drop-down menu to select the **Encoding**, **Sample rate**, **Bit-rate** or **Sample size** parameters. Click **Next** to continue.



8. Click to configure:

- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer Priority**, or
- **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 1000ms depending on the algorithm you select.
- **Local** and **Remote FEC** settings if required.

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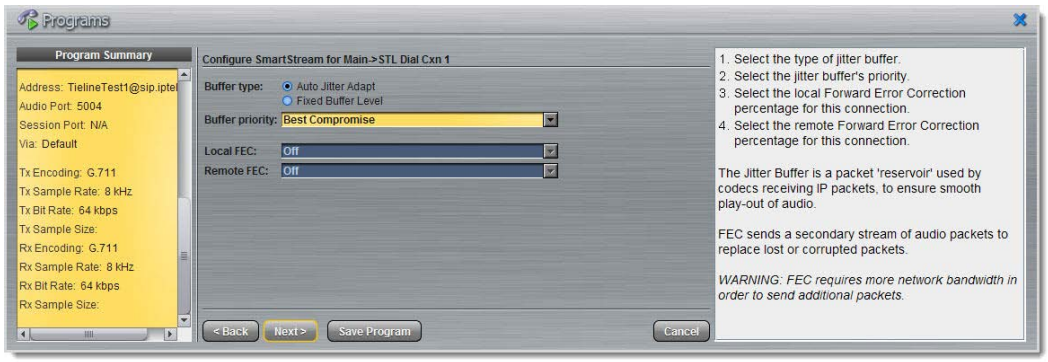
Programs
Program Summary
Address: TielineTest1@sip.iptel
Audio Port: 5004
Session Port: N/A
Via: Default
Tx Encoding: G.711
Tx Sample Rate: 8 kHz
Tx Bit Rate: 64 kbps
Tx Sample Size:
Rx Encoding: G.711
Rx Sample Rate: 8 kHz
Rx Bit Rate: 64 kbps
Rx Sample Size:
Configure SmartStream for Main->STL Dial Cxn 1
Buffer type:
Auto Jitter Adapt
Fixed Buffer Level
Buffer priority:
Best Compromise
Local FEC:
Off
Remote FEC:
Off
< Back
Next >
Save Program
Cancel
1. Select the type of jitter buffer.
2. Select the jitter buffer's priority.
3. Select the local Forward Error Correction percentage for this connection.
4. Select the remote Forward Error Correction percentage for this connection.
The Jitter Buffer is a packet 'reservoir' used by codecs receiving IP packets, to ensure smooth play-out of audio.
FEC sends a secondary stream of audio packets to replace lost or corrupted packets.
WARNING: FEC requires more network bandwidth in order to send additional packets.

Click **Save Program** to save all settings, or click **Next** to configure **Auto Reconnect** or a backup connection using fail over (Note: any backup connection must also use the **ETH1** network port on the codec). If you click **Save Program,** select the check-box if you want to connect the program immediately, then click **Finish**.

Programs
You've successfully created a program. You can connect this program from the Master panel.
Connect this program when I click Finish
< Back
Next >
Finish
Close

9. The newly created program will be displayed in the **Programs panel** and in the **Master panel**. Dial the program by loading and connecting using the **Master panel**, or dial the program manually using the codec front panel.

Caution: If the codec LAN cable is disconnected and the IP address changes when dialing in SIP mode, you will need to reboot the codec to re-register it to your SIP server account. Otherwise the codec will not be able to reconnect.

14.9 Reset Factory Default Settings

There are several settings which allow you to restore factory default settings within the codec.

1. Open the web-GUI and click the **Configure** button at the top of the screen to display the **Configure panel**.

2. Click the **Reset** button at the top of the **System panel**.

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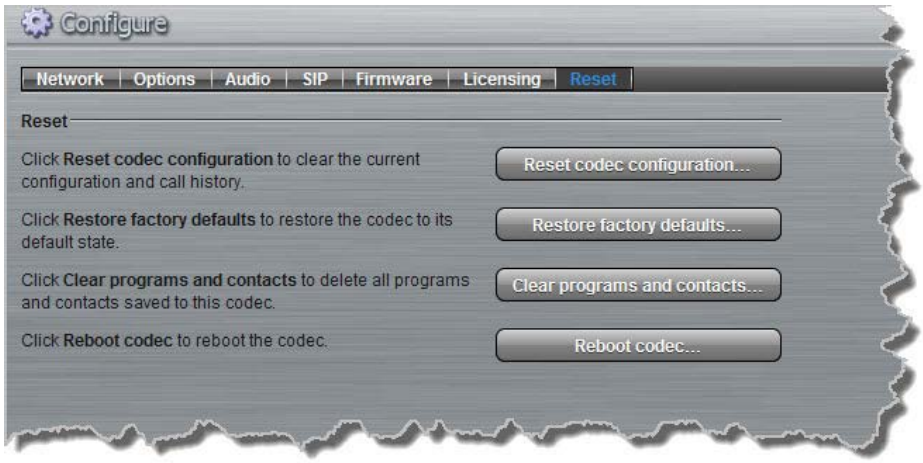
3. Click one of the four reset options available.



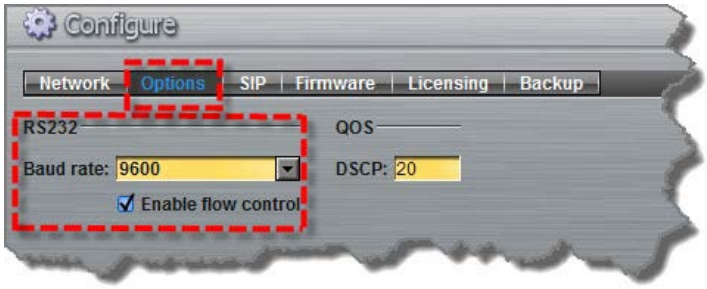
4. A confirmation dialog appears for each option, click **Yes** to proceed or **No** to cancel the reset function.

14.10 RS232 Data Configuration

The codec can be connected to external devices and send RS232-compatible data via the serial port on the rear panel of the codec. To enable RS232 data within a connection, select **Enable Auxiliary Data** when creating a program in the **Programs panel** wizard. Alternatively, select using the codec **Config** menu (see Enabling RS232 Data).

Setting RS232 Data Rates and Flow Control

1. Open the web-GUI and click the **Configure** button at the top of the screen to display the **Configure panel**.
2. Click the **Options** button.
3. Click the **Baud rate** drop-down menu arrow to select the serial port baud rate which matches the baud rate of the external device connected to the RS232 port on the codec.
4. Click to select the **Enable flow control** check box and enable flow control, then click **Save settings**.



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Important Notes:
- The codec cannot send RS232 data or activate relays on IP-enabled Tieline G3 codecs.
- It is important to enable serial port flow control as it regulates the flow of data through the serial port. If disabled, data will flow unregulated and some may be lost.
- Ensure you match the serial port baud rate to match the rate of the external device you are connecting to. Ideally the settings on both codecs should match, or you could have data overflow issues.
- Only the dialing codec needs to be programmed to send RS232 data. Session data sent from the dialing codec will program all other compatible codecs (non-G3) when you connect.
- RS232 data can be sent from the dialing codec to all end-points of a multi-unicast connection if your codec is capable of these connections. Note: Bidirectional RS232 data is only available on the first connection dialed when multi-unicasting.

14.11 Creating Rules

The **Rules panel** in the Toolbox web-GUI is used to program events for specific codec actions. Typically these 'rules' are based on a change in the state of a GPIO control port or the codec being connected or disconnected. Rules can only be created with the web-GUI while the codec is disconnected.

Important Note: Data transmission is disabled by default. **Data** must be enabled in the **Connection** menu to enable contact closure operation and RS232 data.

1. Press the **HOME** button to return to the **Home** screen
2. Use the navigation buttons on the front panel to select **Connect** and press the OK button
3. Select **IP** and press the OK button
4. Use the down navigation button to select **Config** and press the OK button.
5. Navigate to **Data** and press OK to toggle between **Enabled** and **Disabled**.

For more information please see Enabling Relays & RS232 Data.

Programming Rules

Default rules have been preprogrammed into the codec to facilitate programming the most common events required by broadcast engineers. To view rules options:

1. Click the **Rules** button at the top of the web-GUI screen to open the **Rules panel**.
2. Click **Add New Rule**.
3. Click to select the appropriate programming rule for your requirements. See the Web-GUI Introduction section for explanations of the actions each rule can perform.

Rules
What do you want this rule to do?
Connect and disconnect a program when an input is toggled
E.g. Connect "My Program" when Input 1 is switched ON, and disconnect it when Input 1 is switched OFF.
Connect a program when an input is switched ON and disconnect it when another input is switched ON
E.g. Connect "My Program" when Input 1 is switched ON, and disconnect it when Input 2 is switched ON.
Synchronise a local input to a remote relay
E.g. Set remote relay 1 to the value of input 1.
Toggle a relay based on a connection's status
E.g. Switch relay 1 ON when "My Program" is connected, and switch it OFF when "My Program" is disconnected.
Cancel

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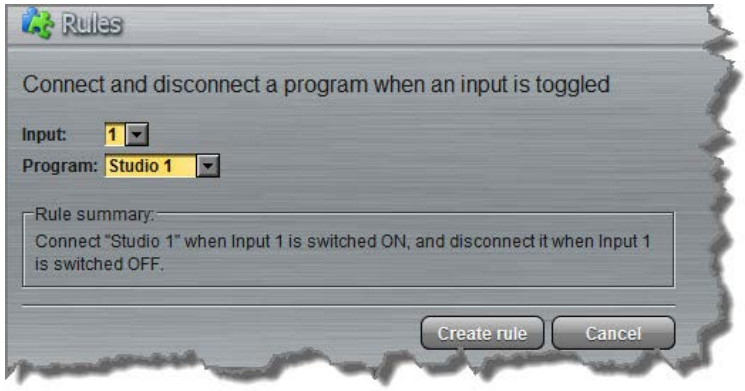
Rule 1: Toggle a Control Port Input to Connect and Disconnect a Program

This rule is used to connect and disconnect a selected program when a control port input is toggled.

1. Click the first rule in the **Rules panel**.
2. Click the drop-down **Input** arrow and select the control port input which will trigger program connection and disconnection.
3. Click the drop-down **Program** arrow to select the program to be connected.

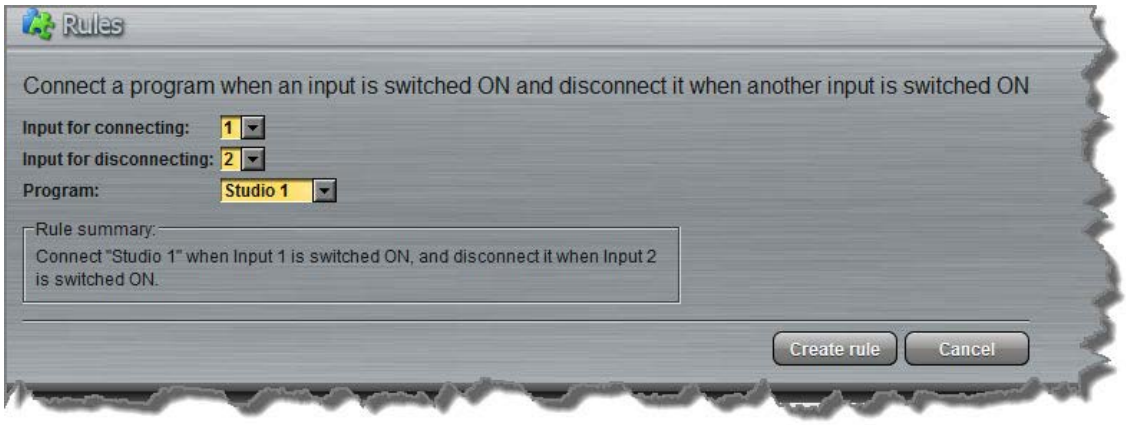

Rules
Connect and disconnect a program when an input is toggled
Input: 1
Program: Studio 1
Rule summary:
Connect "Studio 1" when Input 1 is switched ON, and disconnect it when Input 1 is switched OFF.
Create rule
Cancel

4. Check the **Rule Summary** and click **Create Rule** to save the settings.

Rule 2: Switch Different Control Port Inputs On to Connect and Disconnect a Program

This rule is used to connect and disconnect a selected program when different codec control port inputs are turned on.

1. Click the second rule in the **Rules panel**.
2. Click the drop-down arrows to select the control port input for connecting and the alternative one for disconnecting.
3. Click the drop-down **Program** arrow to select an individual program which will be connected and disconnected by the change in the control port input states.


Rules
Connect a program when an input is switched ON and disconnect it when another input is switched ON
Input for connecting: 1
Input for disconnecting: 2
Program: Studio 1
Rule summary:
Connect "Studio 1" when Input 1 is switched ON, and disconnect it when Input 2 is switched ON.
Create rule
Cancel

4. Check the **Rule summary** and click **Create Rule** to save the settings.

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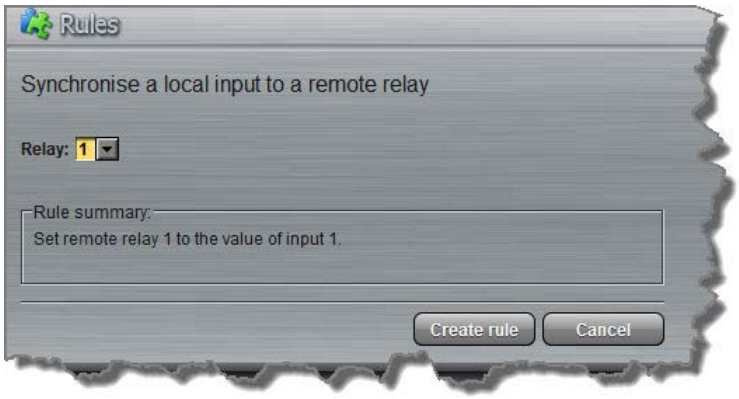
Rule 3: Synchronise Local Control Port Input Status with a Remote Relay Output

Use this rule allow a local codec's control port input to change the state of a remote relay output.

1. Click the third rule in the **Rules panel**.
2. Click the drop-down arrow to select the local control port input used to control a remote relay output.



3. Check the **Rule summary** and click **Create Rule** to save the settings.

Rule 4: Toggle a Relay Output with each Change in Connection Status

This rule is used to toggle a codec's control port relay output each time a program connects and disconnects.

1. Click the fourth rule in the **Rules panel**.
2. Click the drop-down **Relay** arrow and select the relay output you want to toggle.
3. Click the drop-down **Program** arrow to select a specific program which will affect the relay toggle function, or use the default setting whereby any program will toggle the relay output.

Rules
Toggle a relay based on a connection's status
Relay: 1
Program: Studio 2
Rule summary:
Switch relay 1 ON when "Studio 2" is connected, and switch it OFF when "Studio 2" is disconnected.
Create rule
Cancel

4. Check the **Rule summary** and click **Create Rule** to save the settings.

Deleting Rules

1. Click the **Rules** button at the top of the web-GUI screen to open the **Rules panel**.
2. Click the **Delete** button next to the rule you want to delete.
3. Click **Yes** in the confirmation dialog.

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14.12 Upgrading Codec Firmware

Automatic Firmware Upgrades

By default the web-GUI application integrates with TieServer to automatically update users when a firmware upgrade is available.

1. Connect your codec to your PC using either a LAN or USB connection and open the web-GUI program (See Connecting to the Web GUI)
2. If new software is available the **Update** symbol appears in the top-left hand side of the screen.
3. Position your mouse-pointer over the **Update** symbol and click the update dialog when it appears to download the new software.

4. Click **More Information** in the **Updating firmware** dialog to display details of the upgrade process.

Important Note: Firmware upgrade files are very large and it is usually much quicker to download the file to your PC first and then upgrade using the following procedure.

Manual Firmware Upgrades

It is possible to program the web-GUI to allow codec firmware upgrades by selecting a file on a PC.

1. Click the **Configure** button at the top of the web-GUI screen if the **Configure panel** is not displayed.
2. Click **Firmware**.

3. Click **Update from a selected file** and click the **Select File** button.
4. Select the **.bin** file you are using to perform the upgrade and click **Open**.
5. Press the **Update Firmware** button to commence the upgrade.

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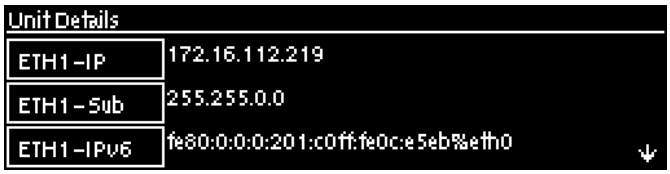
15 Front Panel Configuration Tasks

The codec supports IPv4 and IPv6 IP addressing for connecting over IP networks. It also includes SmartStream IP technology for automatically managing connections over IP networks. This includes advanced Forward Error Correction (FEC) techniques designed to increase the stability of IP connections in the event that data packets are lost, as well as sophisticated automatic jitter buffering that simplifies management of network congestion over a range of IP network infrastructure.

15.1 Configuring IP via the Front Panel

Checking IP Address Details in the Codec

1. Press the **CONFIGURE** button.
2. Select **Unit** and press the OK button.
3. IP address details and other unit details are listed. Use the arrow up and down buttons to scroll and view all details listed.



Important Note: See the Configuring IP Connections sections for more details about IP connections. For assistance with configuration of IPv4 or IPv6 network connections contact your IT Administrator.

Configure an IPv4 DHCP Address

By default the codec is programmed for DHCP-assigned IP addresses. DHCP IP addresses are automatically assigned and can change each time you connect to your Internet Service Provider or by a router on your local area network (LAN).

1. Press the **CONFIGURE** button.
2. Select **LAN** and press the OK button.
3. Use the down navigation button to select **ETH1** or **ETH2**.
4. Select **Usage** and choose the appropriate control and/or streaming mode for the connection, then press the OK button.
5. Select **IPv4 Mode** and press the OK button.
6. Select **DHCP** and press the OK button.
7. Use the up navigation button to scroll to the top of the menu and select **Apply Setting**, then press the OK button to confirm the new settings.

Configure a Static IPv4 Address

Static IP addresses are fixed addresses which are recommended for studio installations. Using a static IP address ensures remote codecs can connect reliably using the same IP address over time.

1. Press the **CONFIGURE** button.

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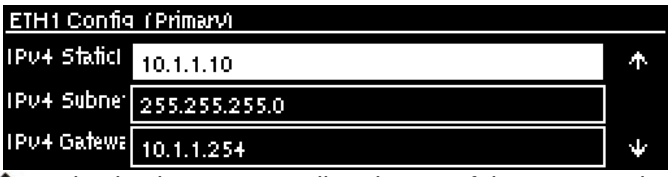
2. Select **LAN** and press the OK button.
3. Use the down navigation button to select **ETH1** or **ETH2**.
4. Select **Usage** and choose the appropriate control and/or streaming mode for the connection, then press the OK button.
5. Select **IPv4 Mode** and press the OK button.
6. Select **Static** and press the OK button.
7. Navigate to **IPv4 Static** and enter the IP address, then press the OK button.
8. Navigate to **IPv4 Subnet** and enter the Subnet Mask, then press the OK button.
9. Navigate to **IPv4 Gateway** and enter the Gateway details, then press the OK button.



10. Use the up navigation button to scroll to the top of the menu and select **Apply Setting**, then press the OK button to confirm the new settings.
11. Check the **Unit Details** menu to ensure the new static IP address has been entered correctly.

IPv6 Address Assignment

There are three IPv6 settings available for each Ethernet port on the codec.

1. Auto: An address is automatically assigned to the codec when you connect the codec to an IPv6 router. This process is similar to how an IPv4 DHCP address is assigned.
2. Manual: Select to manually enter IPv6 address details.
3. Off: Select to ignore IPv6 address details.

Important Note: Select **Off** if you are not using IPv6 to connect to another device. This ensures your codec will attempt to connect using IPv4 at all times.

To adjust this setting:

1. Press the **CONFIGURE** button.
2. Select **LAN** and press the OK button.
3. Use the down navigation button to select **ETH1** or **ETH2**.
4. Select **IPv6 Mode** and press the OK button.
5. Select **Auto**, **Manual** or **Off** and press the OK button.

By default the codec is programmed to allow the codec to automatically receive IPv6 address information from an IPv6 enabled router.

Manual IPv6 Address Assignment

Select **Manual** mode using the previous procedure and enter information into the **IPv6 Static** (Address)**, IPv6 Prefix** and IPv6 **Gateway** fields in the codec to manually program address details.

DNS Server

It is possible to specify Domain Name Server (DNS) settings to allow easy look up of codecs within the specified **DNS Addresses** or **Domains** section within the web-GUI. This feature can be turned

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on or off in the LAN codec menu.

1. Press the **CONFIGURE** button.
2. Use the navigation buttons on the front panel to select **LAN** and press the OK button.
3. Use the down navigation button to select **ETH1** or **ETH2**.
4. Use the arrow down button to scroll to **Auto DNS**.
5. Press the OK button to toggle between **Yes** and **No**.

15.2 Selecting an Algorithm

The codec offers uncompressed audio capability, in addition to apt-X Enhanced, LC-AAC, HE-AAC v.1 and HE-AAC v.2, MPEG Layer 2, G.711 and G.722 algorithms, as well as Tieline Music and MusicPLUS. There is a range of pre-programmed connection profiles that make it very simple to easily program your codec to connect using all available algorithms. See Choosing Dialing Profiles for more details.

Overview of Tieline Algorithms

1. The Tieline Music algorithm is optimized for audio bit rates as low as 19.2kbps with only a 20 millisecond encode delay. It offers 15 kHz mono from 24Kbps to 48Kbps.
2. Tieline MusicPLUS delivers up to 20 kHz mono from 48kbps upwards. It can also deliver up to 20 kHz stereo from 96kbps upwards, offering huge savings on your IP data bills and outstanding audio quality.

Overview of AAC Algorithms

AAC-LC

LC-AAC is optimised for audio bit-rates of 64Kbps per channel or higher using a sample rate of 48kHz. Tieline recommends using LC-AAC instead of HE-AAC if bandwidth of 64Kbps or higher per channel is available, to optimise audio quality. If lower bandwidth than 64Kbps is available consider using HE-AAC, Tieline Music or Tieline MusicPLUS.

AAC-HE

Codecs include both HE-AAC v.1 and HE-AAC v.2, which are optimised for low bit-rate connections. Selection of HE-AAC v.1 and v.2 is automatically managed within the codec, so only **AAC-HE** is displayed on the screen. HE-AAC v.1, when used for mono connections, performs best at bit rates of 24Kbps per channel or higher. HE-AAC v.1 is also used for stereo connections when audio connection bandwidth is 48Kbps or higher.

HE-AAC v.2 is used for stereo connections when audio connection bandwidth is below 48Kbps and is capable of delivering 15kHz quality stereo audio at audio bit-rates as low as 24Kbps.

Overview of apt-X Enhanced Audio Coding

apt-X Enhanced audio coding is used by thousands of radio stations to deliver very low delay audio for IP broadcasts and is ideal for high quality studio-to-transmitter links and audio distribution. It delivers outstanding audio quality with exceptionally low delay across a range of IP networks.

32kHz or 48kHz sample rates are available at either 16 bit or 24 bits per sample. apt-X Enhanced has a minimum connection bit-rate of 128Kbps per channel and offers 10Hz to 24kHz frequency response. 24 bit, 48kHz apt-X Enhanced at the maximum bit-rate of 576Kbps delivers >120dB of dynamic range.

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Programming an Algorithm into the Codec

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the OK button
3. Select **IP** and press the OK button
4. Use the down navigation button to select **Config** and press the OK button.
5. Navigate to **Alg** and press OK.
6. Navigate to **Manual** to configure all settings manually, or **Profile** to choose a pre-configured algorithm profile, then press OK.

How do I Choose an algorithm?

The algorithm you select will not only affect the quality of the broadcast but it will also contribute to the amount of latency or delay introduced. For example, if MP2 algorithms are used, program delays will be much longer than when using Tieline Music or MusicPLUS algorithms. This is due to the additional inherent encoding delays involved when using MP2 algorithms. This can be a major consideration for live applications that integrate remotes into a broadcast. The algorithm you choose to connect with will also depend upon:

- The codecs you are connecting to (Tieline versus non-Tieline)
- Whether you are creating multi-unicast connections.
- Whether you are connecting using SIP or not.
- The uplink bandwidth capability of your broadband connection.

Important Notes: Music and MusicPLUS algorithms cannot be used over SIP connections. Use MP2 algorithms at 64kbps mono or 128kbps stereo for high quality connections when using SIP, or use G.711 and G.722 if required. Tieline G3 codecs do not support connections using AAC and will default to MPEG Layer 2 if an incoming connection is programmed to use this algorithm.

It can be a good idea to listen to the quality of your program signal using each algorithm and to see how it sounds when it is sent at different connection bit-rates (as well as different FEC and jitter-buffer millisecond settings). This will assist you to determine which is the best algorithm setting for the connection you are setting up. Please see the following table for details on the connection requirements of the different algorithms available.

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Algor-ithm	Audio Band-width	Algor-ithmic Delay	IP bit-rate per channel	IP over-head	Recommended connection for on-air use
Linear (Uncom-pressed)	16/24 bit up to 24kHz	0ms	sample rate x bits per sample x no. channels	80Kbps	Extremely high quality uncompressed audio distribution and STLs
Tieline Music	Up to 15kHz	20ms	24 Kbps minimum	16Kbps	High quality low bit-rate remotes, STLs and audio distribution
Tieline Music-PLUS	Up to 22kHz	20ms	48 Kbps minimum	16Kbps	Very high quality low bit-rate remotes, STLs and audio distribution
G.711	3kHz	1ms	64Kbps minimum	80Kbps	Voice quality connections to other brands of audio codec
G.722	7kHz	1ms	64Kbps minimum	80Kbps	Voice quality connections to other brands of audio codec
MPEG Layer 2	Up to 22kHz	24 to 36ms	64Kbps minimum	8.5 - 13.3Kbps	Very high quality audio connections between Tieline or other brands of codec.
MPEG Layer 3	Up to 15kHz	100ms	64Kbps	8.5 - 13.3Kbps	High quality low bit-rate remotes, STLs and audio distribution
LC-AAC	Up to 15kHz	64ms	64Kbps	15Kbps	High quality low bit-rate remotes, STLs and audio distribution
HE-AAC v.1	Up to 15kHz	128ms	48Kbps	7.4Kbps	High quality low bit-rate remotes, STLs and audio distribution
HE-AAC v.2	Up to 15kHz	128ms	Minimum 16Kbps (Mono); 24Kbps (stereo)	7.4Kbps	DAB+ radio streaming and high quality low bit-rate remotes, STLs and audio distribution
apt-X Enhanced	10Hz-24kHz	2.5ms at 48kHz	128Kbps minimum (16bit; 32kHz) to 288Kbps (24bit;48kHz)	80Kbps	Very high quality STLs and audio distribution

15.3 Configuring the Jitter Buffer

A jitter buffer is a temporary storage buffer used to capture incoming data packets. It is used in packet-based networks to ensure the continuity of audio streams by smoothing out packet arrival times during periods of network congestion. Data packets travel independently and arrival times can vary greatly depending on network congestion and the type of network used, i.e. LAN versus wireless networks. The jitter-buffer is encompassed with Tieline's SmartStream IP technology which can:

- Remove duplicate packets.
- Re-order packets if they arrive out-of-order.
- Repair the stream in the event of packet loss (error concealment).
- Manage delay dynamically based on current network congestion.
- Manage forward error correction (FEC).

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Tieline codecs can be used to program either a fixed or automatic jitter buffer and the setting you use depends on the IP network you are connecting over. Over LANs, WANs and wireless networks the automatic jitter buffer generally works well. It adapts automatically to the prevailing IP network conditions to provide continuity of audio streaming and minimises delay.

A fixed jitter buffer is preferable over satellite connections to ensure continuity of signals.

CAUTION: If a Tieline codec connects to a device that is using non-compliant RTP streams then the last fixed setting programmed into the codec will be enabled (default is 500ms). Non-compliant devices include some other brands of codec, web streams and other devices.

Tieline 'Auto Jitter Buffer' Settings

Least Delay: This setting attempts to reduce the jitter buffer to the lowest possible point, while still trying to capture the majority of data packets and keep audio quality at a reasonable level. This setting is the most aggressive in its adaptation to prevailing conditions, so jitter buffer may vary more quickly than with the other settings. It is not recommended in situations where jitter variation is significant and/or peaky. (E.g. 3G/multi-user wireless networks). It is best for stable and reliable links such as dedicated or lightly-loaded WAN/LANs.

Highest Quality: This setting is the most conservative in terms of adapting down to reduce delay. The jitter-buffer setting will actually stay high for a longer period after a jitter spike is detected – just in case there are more spikes to follow. This setting is best used where audio quality is most highly desired and delay is not so critical. Unless delay is irrelevant, this setting is also not recommended over peaky jitter networks (such as 3G) and is best used on more stable networks where large jitter peaks are not as common.

Best Compromise: This (default) setting is literally the midpoint between the jitter buffer levels that would have been chosen for the Highest Quality and Least Delay settings. It is designed to provide the safest level of good audio quality without introducing too much extra delay.

Good Quality and **Less Delay**: These two settings lie between the mid-point setting of Best Compromise and two settings Highest Quality and Least Delay. They indicate a slight preference and may assist in achieving better performance from a connection without incurring extreme delays in transmission or packet loss.

Which Algorithms can use Automatic Jitter Buffering?

The following table provides an overview of which algorithms are capable of using the automatic jitter buffer feature over SIP and non-SIP connections.

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Algorithm	Non-SIP Connections	SIP Connections
Linear (Uncompressed)	✗	✗
Tieline Music	✓	✗
Tieline MusicPLUS	✓	✗
G.711	✗	✓
G.722	✗	✓
MPEG Layer 2	✓	✓
MPEG Layer 3	✓	✗
LC-AAC	✓	✓
HE-AAC v.1	✓	✓
HE-AAC v.2	✓	✓
apt-X Enhanced	✗	✗

Programming Automatic Jitter Buffering (Default Setting)

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the OK button
3. Select **IP** and press the OK button
4. Use the down navigation button to select **Config** and press the OK button.
5. Navigate to **JitBuf** and press OK.
6. Select **Auto Jitter Adapt** and press OK.
7. Select your preferred jitter buffer setting and press OK.

How to get the Best Jitter Buffer Results

When programming automatic jitter buffer settings, establish the IP connection for a while before 'going live', to let the codec evaluate the prevailing network conditions. The initial jitter buffer setting when a codec connects is 500ms and it is kept at this level for the first minute of connection (as long as observed delay values are lower than this point).

After the initial connection period the jitter buffer is adapted to suit the current network conditions and is usually reduced. Establish a connection for at least 5 minutes prior to broadcasting, so that the codec has been provided with enough jitter history to ensure a reliable connection.

There are five states or stages that jitter buffer may display and these can be observed in the connection status screen by pressing the Enter/Dial button while connected. The first four stages are observed in "auto" jitter buffer mode.

1. **Stabilization period (a1):** A few seconds at the start of a connection where no action is taken at all while the establishment of a stable connection means analysis of jitter data is not valid.
2. **Stage 2 (a2):** A compatibility check to ensure the RTP connection is compliant and RTP clocks are synchronized enough to perform jitter analysis.
3. **Stage 3 (a3):** If the compatibility check is successful, this is the analysis hold-off period. During a minute, the jitter buffer is held at a safe, fixed value of 500ms while enough history is recorded to start jitter buffer adaptation.
4. **Stage 4 "live" (A)**: This is where the codec determines it is safe enough to start broadcasting using the auto-jitter buffer level. We recommend running the codec for a few

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more minutes to obtain a more comprehensive history of the connection's characteristics.

5. **Fixed (F):** This state is displayed if the jitter buffer is fixed.

Important Note: The automatic jitter buffer setting can be changed while connected if IP network conditions and audio output quality dictate a change is required. Automatic jitter buffering is disabled for a Raw (linear uncompressed) audio connection.

Auto Jitter Buffer and Forward Error Correction (FEC)

If forward error correction is programmed then additional data packets are sent over a connection to replace any data packets lost. There is no need to modify jitter buffer settings if you are sending FEC data, only if you are receiving FEC data.

The jitter buffer depth on the receive codec needs to be increased if forward error correction is employed. We recommend you add 100ms to the jitter buffer on a codec receiving FEC at a setting of 20% and 20ms at a setting of 100%.

Tieline's auto jitter buffer detects the amount of FEC that is being used and automatically compensates to increase the codec jitter buffer if FEC is being used.

Fixing Jitter Buffer Settings

The default jitter-buffer setting in Tieline codecs is 500 milliseconds. This is a very reliable setting that will work for just about all connections. However, this is quite a long delay and we recommend that when you set up an IP connection you test how low you can set the jitter-buffer in your codec.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the OK button.
3. Select **IP** and press the OK button.
4. Use the down navigation button to select **Config** and press the OK button.
5. Navigate to **JitBuf** and press OK.
6. Select **Fixed Buffer Level** and press OK.
7. Use the numeric **KEYPAD** to enter the fixed buffer value in milliseconds and press OK.

If you change the jitter buffer setting in a codec it will only adjust to the new level when link quality is high (e.g. above 70%). This is done to ensure audio quality is not compromised. When manually programming the jitter-buffer delay in a codec it is necessary to think carefully about the type of connection you will be using. Following is a table displaying rule of thumb settings for programming jitter-buffer delays into your codec.

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Connection	Jitter-Buffer Recommendation
Private LAN	60 milliseconds
Local	100 - 200 milliseconds
National	100 - 300 milliseconds
International	100 – 400 milliseconds
Wireless Network	250 - 750 milliseconds
Satellite IP	500 - 999 milliseconds

Important Note: The preceding table assumes the use of either Tieline Music or Voice G3 algorithms. Do not use Raw (uncompressed) audio over highly contended DSL/ADSL connections without enough bandwidth to support the high connection bit-rates required.

15.4 Configuring Forward Error Correction

Forward Error Correction (FEC) is designed to increase the stability of UDP/IP connections in the event that data packets are lost. FEC works by sending a secondary stream of audio packets over a connection so that if your primary audio stream packets are lost or corrupted, then packets from the secondary stream can be substituted to replace them. The amount of FEC required depends on the number of data packets lost over the IP connection.

Both the local and remote codec FEC settings can be configured in your codec before dialing. These settings can also be changed 'on the run' while the codecs are connected. FEC should only be used if link quality displayed on the codec is below **L:99 R:99**, as it is of no benefit otherwise.

Programming FEC into the Codec

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the OK button
3. Select **IP** and press the OK button
4. Use the down navigation button to select **Config** and press the OK button.
5. Navigate to **FEC** and press OK.
6. Select the local codec FEC setting in the **Local FEC** screen and press OK.
7. Select the remote codec FEC setting in the **Remote FEC** screen and press OK.

The four FEC settings in Tieline codecs are outlined in the following table with their bit rate ratios.

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FEC Setting	Bit rate Ratios	Connection Use
100% (Lowest delay)	A simultaneous dual-redundant stream (1:1 ratio) is sent from the codec. Twice the connection bit rate is required to operate the codec using the 100% setting. E.g. if your connection is 14,400Kbps, you will require an additional 14,400 Kbps of bandwidth to allow for the FEC data stream.	Recommended to be used over wireless and international connections.
50%	Additional data is sent by FEC in a ratio of 2:1.	Recommended for international & national connections
33%	Additional data is sent by FEC in a ratio of 3:1.	Recommended for national and local connections.
20% (Highest delay)	Additional data is sent by FEC in a ratio of 5:1.	Recommended for local and LAN connections.
Off	FEC is off in the codec and the connection bandwidth is equal to the connection bit rate setting in the codec.	Recommended for wired LAN connections & managed T1 & E1 connections for STLs that have connections that aren't shared & have quality of service (QoS).

Important Note: FEC can only be programmed for use with the Music and MusicPLUS algorithms.

How does FEC work?

If you program a FEC setting of 20% and you are losing one packet in every five sent, the lost packet will be replaced by FEC to maintain the quality of the connection. If you are losing more packets than this, say one in three, it will be necessary to increase the FEC setting to 33% to compensate.

Note: There is an inverse relationship between FEC settings and the jitter-buffer millisecond setting that you use for IP connections.

So why not use 100% FEC every time? The answer is because you need twice the bit rate to achieve full redundancy and depending on the link conditions, this could potentially cause more dropouts because of network congestion than it fixes. Here is a simple rule to remember: Your maximum uplink speed is all the bandwidth you have to play with. As a rule of thumb, try not to exceed more than 80% of your maximum bandwidth. If your link is shared, be even more conservative.

You should also consider the remote end too. What is their maximum upload speed? Is the connection shared at either end? Your bit rates, FEC settings and buffer rates must be pre-configured at both ends before you connect, so it's always better to set your connection speed and balance your FEC according to the available uplink bandwidth at each end for best performance.

As an example, if you want 15 kHz mono (using the Tieline Music Algorithm) you will need at least a 24kbps connection for audio. Adding 100% FEC will add another 24kbps making your bit rate 48kbps plus some overhead of around 10kbps is required. If you're on a 64kbps uplink, you should

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consider reducing your FEC to minimise the likelihood of exceeding your bandwidth capacity.

Here is another example, if you want 15 kHz stereo, you need at least 56kbps for the audio. 100% FEC requires at least 112kbps and 50% FEC requires at least 84kbps. If your uplink speed is 256kbps and you're on a shared connection, then choosing a lower FEC setting of 20%-33% may give you better results.

15.5 Configuring Encode/Decode Direction

By default the codec by is configured to both encode and decode data. However, it is possible to configure the codec to either encode or decode audio data only. This is useful for:

- Conserving connection bandwidth when unidirectional data streaming is required.
- Lowering data costs.
- Increasing overall connection reliability.

Program the transmitting codec to encode only and program the receive codec to decode only when using this feature. To adjust this setting:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the OK button
3. Select **IP** and press OK.
4. Use the down navigation button to select **Config** and press OK.
5. Navigate to **Dir** and press OK.
6. Select the encode or decode direction setting you want and press OK.

15.6 Enabling Relays & RS232 Data

Data must be enabled to activate contact **CONTROL PORT** closure operation and RS232 data.

1. Press the **HOME** button to return to the **Home** screen
2. Use the navigation buttons on the front panel to select **Connect** and press the OK button
3. Select **IP** and press the OK button
4. Use the down navigation button to select **Config** and press the OK button.
5. Navigate to **Data** and press OK to toggle between **Enabled** and **Disabled** (Note: default setting is **Disabled**)

Important Note: Data transmission is disabled by default.

Configuring Control Port Contact Closure Operation

The **Rules panel** on the web-GUI can be used to configure switch inputs and relay outputs. See the section titled Creating Rules for more information.

Configuring RS232 Data

Once **Data** is enabled, the codec can be connected to external devices and transport RS232-compatible data via the serial port on the rear panel of the codec.

1. Press the **CONFIGURE** button.
2. Navigate to **System** and press OK.

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3. Select **RS232 Cfg** and press OK.
4. Use the navigation buttons to select the correct baud rate.
5. Select **Enable** for flow control and press OK to save all settings.

Important Notes:

- The codec cannot send RS232 data or activate relays on Tieline G3 codecs.
- It is important that you enable serial port flow control within the codec. Flow control regulates the flow of data through the serial port. If disabled, data will flow unregulated and some may be lost.
- Ensure you match the serial port baud rate to match the rate of the external device you are connecting to. Ideally the settings on both codecs should match, or you could have data overflow issues.
- Only the dialing codec needs to be programmed to send RS232 data. Session data sent from the dialing codec will program all other compatible codecs (non-G3) when you connect.
- RS232 data can be sent from the dialing codec to all end-points of a multi-unicast connection if your codec is capable of these connections. Note: Bidirectional RS232 data is only available on the first connection dialed when multi-unicasting.

15.7 Configuring TCP/UDP Protocols

In TCP and UDP networks the codec port is the endpoint of your connection. Software network ports are doorways for systems to communicate with each other. For example, several codecs in your studio may use the same public static IP address. Therefore it is necessary to allocate port numbers to these codecs, so that when an incoming call comes in, the network knows which codec to send the call to.

Tieline Codec Default Port Settings

By default, the codec uses a TCP session port to send session data and a UDP port to send audio. The session port is programmed to use the TCP protocol because it is the most likely protocol to get through firewalls – ensuring critical session data (including dial, connect and hang-up data) will be received reliably.

When the dialing codec initiates a call, the session data from that codec configures both the local and remote codec session ports, as well as the audio ports, with the settings on the dialing codec. The default session and audio port settings in Tieline codecs, for both TCP and UDP connections, are outlined in the following table.

IP Connection	Session Data Port	Audio Port
IP1 connection	TCP 9002 (to send session data)	UDP 9000 (to send audio data)
Toolbox Web-GUI	TCP 80	
SIP	UDP 5060	UDP 5004

Note: Using a port scanner to test a codec will be unsuccessful if you try to scan and the port is already in use, i.e. the codec is connected.

Changing Codec Port Numbers

Reasons for adjusting the port setting on your codec include:

- Creating a path through gateways and firewalls.

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- Another IP device is already using a codec's port number.
- More than one studio codec is in use and each codec requires a different port number.

For two codecs to connect, they need to be programmed with matching port numbers. If there is a need to change codec port settings, in most situations you should consult your organization's resident IT professional. To adjust either the session or audio port numbers:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Connect** and press the OK button.
3. Select **IP** and press the OK button.
4. Use the down navigation button to select **Config** and press the OK button.
5. Navigate to either **Sess** (session protocol) or **Proto** (audio protocol) and press OK.
6. Select the session or audio protocol you want and and press OK.
7. Use the numeric **KEYPAD** to add a new port number and press OK.

15.8 Configuring QoS for Broadcasts

It is possible for IP networks to prioritise and differentiate between data packets transmitted through routers across networks. This is useful because in modern data networks many different IP services like email, voice, web pages, video and streaming music coexist within the same network infrastructure.

Prioritising IP Data Packets when Broadcasting

IP audio data packets can be programmed for expedited or assured forwarding (Quality of Service or QoS) when traversing different networks. Routers can also be programmed to ignore these forwarding priorities so they are not assured across all networks.

The codec can be programmed to tag IP data packets sent across a network by entering a value into the Differentiated Services Code Point (DSCP) field within the header of data packets transmitted over the network. Check with your IT administrator before changing this setting. By default the codec is programmed for Assured Forwarding and more details about DSCP are available on Wikipedia at http://en.wikipedia.org/wiki/Dscp.

Configuring QoS

1. Press the **CONFIGURE** button.
2. Use the navigation buttons to select **QoS** and press the OK button.

3. Press the OK button and use the **RETURN** button to delete numbers already entered, then use the numeric **KEYPAD** to enter the new setting recommended by your IT administrator.
4. Press the OK button to save the new setting.

Important Note: To ensure the continuous and regular flow of tagged data packets along the path from point to point, all routers and switching equipment must respect the QoS

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setting of the packets sent. Any bandwidth partitioning schemes should partition over a small interval to ensure the codec jitter buffer does not empty and audio remains continuous.

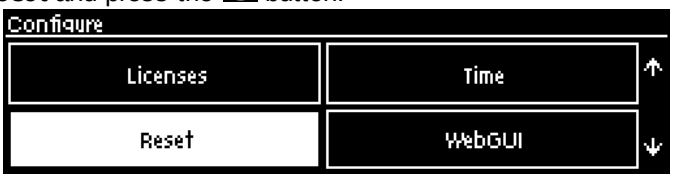
15.9 Reset and Restore Factory Default Settings

Several reset and reboot functions are available from within the **Reset Functions** menu. These options include:

	Function	Description
1	Reset Codec Config	This clears all codec settings but retains key settings that include LAN, language, LCD contrast and contact settings
2	Restore Factory Defaults	This clears all codec settings back to their factory defaults
3	Clear Programs & Recent Calls	This clears the list of programs and recent calls in the codec, buddy list contacts are retained
4	Reboot Codec	Reboots the codec
5	Clear Logs	Clears logs from the codec

Important Note: After restoring factory defaults, always reboot the codec using the **Reboot Codec** function, not by removing power from the codec.

1. Press the **CONFIGURE** button.
2. Navigate to **Reset** and press the OK button.



3. Navigate to the preferred option from those available and press the OK button.
4. Select **Yes** and press the OK button to confirm the reset function.

Reset and Restore Factory Defaults using the Web-GUI

The web-GUI can also be used to reset and restore factory defaults. See Reset Factory Default Settings for more details.

15.10 Test Mode

Test mode is used by the codec to perform an input/output loopback test of audio. E.g. channel 1 is routed to channel 1 out.

1. Press the **CONFIGURE** button.
2. Navigate to **Audio** and press OK.
3. Navigate to **Test Mode** and press OK

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16 Reference

The following sections contain reference and troubleshooting information.

16.1 Regular Maintenance

Tieline recommends the codec undergoes regular maintenance to ensure operational efficiency and prolong its life.

WARNINGS: All work should be carried out by suitably qualified personnel. Remove both power leads from the codec before removing the cover. All parts are mounted on plugs and only a Philips screwdriver is required. Ensure that fan mounting lugs are not hooked out by the cover.

Maintenance Schedule

Tieline recommends a three year maintenance schedule which includes the following procedures to be completed:

1. Evacuate all dust from the unit and clean vents.
2. Replace both PSUs.
3. Replace the fan.

Controlled rack environments may allow a longer maintenance cycle. Uncontrolled environments, where temperatures are elevated, may require a shorter maintenance cycle.

Tieline recommends that the racks in which codecs are installed are thoroughly evacuated to ensure proper airflow from the bottom to the top. Where space is available, a 1RU gap between codecs will assist in minimising internal temperature build up. Tieline has incorporated dual redundant PSUs and backup alarm features to assist in maintaining reliable operations. The fan has been carefully chosen for long life operation and should not be replaced by a cheaper equivalent. Fan speed control circuitry reduces the fan speed as internal rack temperatures fall below 25 degrees Celsius. This greatly extends the working life of the fan and the codec. If rack temperatures are elevated above 25 degrees Celsius, the fan speed will increase to reduce CPU temperature.

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16.2 Tips for Creating Reliable IP Connections

The following 10 tips are provided to help obtain the best possible IP connection between two codecs, without paying for Quality of Service (QoS).

1. Always use the best quality Internet Service Provider (ISP). Tier 1 service providers are best as their infrastructure actually makes up the internet 'backbone'. Wikipedia lists the major service providers that make up the internet backbone at: http://en.wikipedia.org/wiki/internet_backbone. In Australia Telstra is equivalent to one of these service providers.

2. You will get the best quality connection if both the local (studio) and remote codecs use the same ISP. This can substantially increase reliability, audio bandwidth and reduce audio delay. Using the same service provider nationally can give better results than using different local service providers. This is especially true if one of the service providers is a cheap, low-end domestic service provider, which buys its bandwidth from other ISPs. Second and third tier providers sublease bandwidth from first tier providers and can result in connection reliability issues due to multiple switch hops. We also highly recommend using First Tier ISPs if connecting two codecs in different countries.

3. Sign up for a business plan that provides better performance than domestic or residential plans. Business plans typically have a fixed data limit per month with an additional cost for data beyond that limit. In addition, Service Level Agreements (SLA) will often provide better support and response times in the event of a connection failure. Domestic plans are often speed-limited or "shaped" when usage exceeds a predefined limit. These plans are cheap but they are dangerous for streaming broadcast audio.

4. Ensure that the speed of the connection for both codecs is adequate for the job. The minimum upload speed recommended is 256 kbps for a studio codec and 64 kbps for a field unit connection.

5. Use good quality equipment to connect your codecs to the internet. (Tieline successfully uses Cisco® switching and routing equipment.):

- If you are using a DSL or ADSL connection make sure you purchase a high quality modem that can easily meet your speed requirements. This is especially important if you are over 4 kms from an exchange.
- If you have multiple codecs connected to a local area network (LAN) please ensure that your network infrastructure is designed for media streaming and not domestic usage. Tieline has tested several cheap 8-port switches that lose more packets between local computers than an international IP connection between Australia and the USA!
- If using a wireless connection ensure that the antenna signal strength received is strong. The type of antenna used and the amount of output gain also affects connection quality.

Important Note: You should be able to stream audio between two codecs on your LAN and get 'link quality' readings of L99R99. If you see anything less than this then you should get a network engineer to investigate the issue.

6. Once your internet connection is installed at the studio check that the connection performance is approximately what you ordered and are paying for. A connection can perform below advertised bit rates if:

- There is an error in ISP configuration;
- There is an error in modem configuration;
- There is a poor quality line between the studio and the exchange;
- There are too may phones or faxes connected to the phone line; or
- Line filters have been connected incorrectly.

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Tech Note: You can test your internet connection speed by connecting a PC to the internet and using http://www.speedtest.net/index.php . If the bandwidth detected is low then something is wrong. Get it fixed before going live!

7. Use a dedicated DSL/ADSL line for your codecs. Do not share a link with PCs or company networks. The only exception to this rule is if an organisation has network equipment and engineers that can implement and manage quality of service (QoS) on its network.

8. Use UDP as the preferred audio transport protocol. TCP generally results in lower bit rates and random drop-outs of audio over the internet. Only use TCP if UDP is blocked by firewalls and you are unable to connect.

9. When using UDP ensure the total bit rate (audio bit rate plus header bit rate) is no more than 80% of the ISP connection rate. IP headers require around 20 kbps in addition to the audio bit rate. For example, with a 64 kbps connection the audio bit rate should be (64-20) x 0.8 = 31.2 kbps or lower. For TCP we suggest a limit of 50% or less.

10. Wireless IP connections can easily become congested and result in packet loss and audio drop-outs. It is very difficult to guarantee connection quality when there is no way of knowing how many people are sharing the same wireless connection.

Important Note: Be careful when using cell-phone connections at special events where thousands of people have mobile phones. This can result in poor quality connections and audio drop-outs if cell-phone base stations are overloaded.

IP Connection Checklist

Complete the following check list and aim for a score of at least 8 out of 10 before going live.

Number	Check	Result
1	Using a reputable Tier1 ISP that's part of internet backbone.	
2	The same ISP is being used for both codec connections.	
3	The ISP Plan is a Business Plan or equivalent.	
4	The ISP connection speed is adequate.	
5	Equipment is high quality and suitable for media streaming.	
6	The ISP connection speed has been tested and is suitable.	
7	The ISP connection is not shared with other PCs or devices.	
8	UDP is being used as the audio transport protocol.	
9	No more than 80% of ISP connection bandwidth is being used.	
10	There are no wireless connections being used.	

16.3 Merlin Compliances and Certifications

Declaration of Conformity

This Merlin codec meets the requirements of directives for CE and C-Tick certifications. Technical documentation required by the conformity assessment procedure is kept at the head office of Tieline Technology; 1/25 Irvine Drive, Malaga, Western Australia 6090.

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EN 55 022 Statement

This is to certify that Tieline Merlin is shielded against the generation of radio interference in accordance with the application of EN 55 022: 2006 Class A. Technical documentation required by the conformity assessment procedure is kept at the head office of Tieline Technology; 1/25 Irvine Drive, Malaga, Western Australia 6090.

Canadian Department of Communications Radio Interference Regulations

This digital apparatus (Tieline Merlin) does not exceed the Class B limits for radio-noise emissions from digital apparatus as set out in the Radio Interference Regulations of the Canadian Department of Communications.

Règlement sur le brouillage radioélectrique du ministère des Communications

Cet appareil numérique (Tieline Merlin) respecte les limites de bruits radioélectriques visant les appareils numériques de classe B prescrites dans le Règlement sur le brouillage radioelectrique du ministère des Communications du Canada.

16.4 FCC Compliance Statements

FCC Part 15

Compliance: TIELINE PTY LTD, 25 Irvine Drive, Malaga. Western Australia 6090.

This equipment has been tested and found to comply with the limits for a class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area may cause harmful interference, in which case the user will be required to correct the interference at his/her own expense. Changes or modifications not expressly approved by Tieline Pty Ltd could void the user's authority to operate the equipment.

If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try and correct the problem by one or more of the following measures:

1. Increase the separation between the equipment and the receiver;
2. Connect the equipment into an outlet on a circuit different to that used by the receiver;
3. Consult the dealer or an experienced radio/TV technician.

FCC Part 68

FCC Registration Number: 6NAAUS-34641-MD-E
Ringer Equivalence Number (REN):0.5B

A label containing, among other information, the FCC registration and Ringer Equivalence Number (REN) for this equipment is prominently posted on the bottom, near the rear of the equipment. If requested, this information must be provided to your telephone company. USOC Jacks: This device uses RJ11C terminal jacks. The REN is used to determine the quantity of devices, which may be connected to the telephone line. Excessive RENs on the telephone line may result in the devices

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not ringing in response to an incoming call. In most, but not all areas, the sum of RENs should not exceed five (5). To be certain of the number of devices that may be connected to the line, as determined by the total RENs, contact the telephone company to obtain the maximum RENs for the calling area.

If this equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of the service may be required. If advance notice is not practical, the company will notify the customer as soon as possible. Also you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The Telephone Company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens the Telephone Company will provide advance notice for you to make the necessary modifications in order to maintain uninterrupted service.

If you experience problems with this equipment, contact TIELINE Pty Ltd, 25 Irvine Drive, Malaga. Western Australia, 6090. Ph +61 8 9249 6688 Fax +61 8 9249 6858 email info@tieline.com (web page www.tieline.com) for repair and warranty information.

If the problem is causing harm to the telephone network, the Telephone Company may request you remove the equipment from the network until the problem is resolved.

No user serviceable parts are contained in this product. If damage or malfunction occurs, contact TIELINE Pty Ltd for instructions on repair or return. This equipment cannot be used on a telephone company provided coin service. Connection to Party Line service is subject to state tariffs.

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17 Merlin Specifications

Input/Output Specifications	
Analog Audio Inputs	2 x Female XLR line inputs
Analog Audio Outputs	2 x Male XLR
AES3 In	1 x female XLR (Channel 1 in; shared with Ch1 analog input)
AES3 Out	1 x male XLR
Aux Input	1 x 6.35mm (1/4") Mic/Line level Jack on rear panel
Headphones Out/ Aux Out	1 x 6.35mm (1/4") Jack on rear panel and 1 x 6.35mm (1/4") Jack on the front panel
Control Port In/Out	Four relay inputs and four opto-isolated outputs for machine control via a DB15 connector.
Audio Input Impedance	High Impedance > 5K ohm
Output Impedance	<50 ohm Balanced
Clipping Level	+22dBu (input and outputs)
A/D & D/A Converters	24 bit
Frequency Response	20Hz to 22kHz
Total Harmonic Distortion	<0.03% at +4dBu, unweighted
Signal To Noise Ratio	>90dB at +22dBu, unweighted
Sample Frequencies	
IP Sample Frequencies	16kHz, 24kHz, 32kHz, 48kHz
Algorithms	
IP	Tieline Music, Tieline MusicPLUS, G.711, G.722, MPEG Layer 2
IP	LC-AAC, HE-AAC and HE-AACv2
IP	16/24 bit Enhanced apt-X
IP (uncompressed)	Linear PCM16/24 bit 48kHz sampling
Data and Control Interfaces	
USB	USB 2.0 Host port on the front panel
LAN	2 x 10/100/1000 RJ45 connectors
Serial	RS232 up to 115kbps with or without CTS/RTS flow control via female DB9 connector, can be used as a proprietary data channel
Front Panel Interfaces	
Display	256 x 64 monochrome LCD
Keypad	21 button keypad
Navigation	5 button keypad
General	
Size	1U x 19" Rackmount
Dimensions	18.98” x 1.73” x 13.5” [482mm (W) x 44mm (H) x 343mm (D) including rear connectors]
Weight	6lb 3.47oz/2.82Kg
Power Consumption	Dual 100-240V AC power inputs
Operating Temperature	0°C to 45°C (40°F to 113°F)
Humidity Operating Range	0 to 70% (at 35°C non-condensing)

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18 Appendix A: RS232 and Control Port Wiring

Relays

The codec uses a DB15 connector to facilitate use of four CMOS solid state relays for the control of equipment, consisting of four relay closures and four opto-isolated outputs.

Inputs

The input signal is referenced to chassis ground, i.e. the ground reference terminal on the connector is connected the chassis. The input device is a high impedance CMOS device with a 330 ohm pull-up resistor to +5 volts.

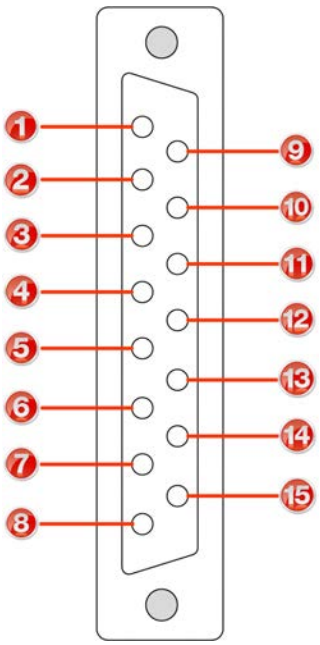
Operation is as simple as joining the input pin to the ground terminal. This can be via a remote relay contact or the open circuit collector of a transistor or FET. DO NOT feed voltages into the inputs.

Outputs

CMOS field effect transistors switch a low impedance path between the two pins when activated. These are opto-isolated and floating above ground. It is important to current-limit the source as damage will result where the current exceeds 100mA peak-to-peak. No more than 48 volts peak-to-peak should be used as a safety precaution. The resistance of the CMOS element is approximately 25 ohms in the ON state.

Control Port Pin-outs

A closing contact across Inputs 1-4 to Ground will provide a closing contact on the remote codec Outputs 1 to 4. If your codec supports multi-unicast connections to multiple codecs, a contact closure will appear on each of the compatible (non-G3) remote codecs' corresponding contacts. I.e. Input 1 shorted, Output 1 contacts on all connected codecs closed.



Female DB-15
Codec Connector

Pins	Pin Function
1	Ground
2	Output 4
3	Output 3
4	Output 2
5	Output 1
6	Ground
7	Input 3
8	Input 1
9	Output 4
10	Output 3
11	Output 2
12	Output1
13	Ground
14	Input 4
15	Input 2

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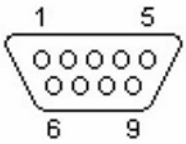
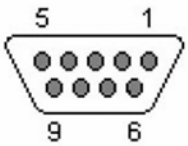
Important Note: For more information about how to program relay operations with a PC using the Toolbox web-GUI, please see Creating Rules.

RS232 Pin-outs and Data Connections

Pin	INTERFACE Female DB9 (RS232) DCE	DATA Male DB9 (RS232) DTE
1	No Connection	No connection
2	TX Data	RX Data
3	RX Data	TX Data
4	No connection	No connection
5	Signal Ground	Signal Ground
6	No Connection	No connection
7	CTS	RTS
8	RTS	CTS
9	No connection	No connection



DB9 Male Connector Pins



DB9 Female Connector Pins

Important Notes:

- The codec cannot send RS232 data to, or activate relays on Tieline G3 codecs.
- It is important that you enable serial port flow control within the codec. Flow control regulates the flow of data through the serial port. If disabled, data will flow unregulated and some may be lost.
- Ensure you match the serial port baud rate to match the rate of the external device you are connecting to. Ideally the settings on both codecs should match, or you could have data overflow issues.
- Only the dialing codec needs to be programmed to send RS232 data. Session data sent from the dialing codec will program all other compatible codecs (non-G3) when you connect.
- RS232 data can be sent from the dialing codec to all end-points of a multi-unicast connection if your codec is capable of these connections. Note: Bidirectional RS232 data is only available on the first connection dialed when multi-unicasting.

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